

SEARCH REQUEST FORM

Scientific and Technical Information Center

Requester's Full Name: Fan Tsang Examiner #: _____ Date: 9/26/00
 Art Unit: 2746 Phone Number 305 4895 Serial Number: 08 948 328
 Mail Box and Bldg/Room Location: 8221 Results Format Preferred (circle): PAPER DISK E-MAIL

If more than one search is submitted, please prioritize searches in order of need.

Please provide a detailed statement of the search topic, and describe as specifically as possible the subject matter to be searched. Include the elected species or structures, keywords, synonyms, acronyms, and registry numbers, and combine with the concept or utility of the invention. Define any terms that may have a special meaning. Give examples or relevant citations, authors, etc, if known. Please attach a copy of the cover sheet, pertinent claims, and abstract.

Title of Invention: Personal Message Service with Enhanced Text To Speech Synthesis

Inventors (please provide full names): _____

Earliest Priority Filing Date: 10/10/97

For Sequence Searches Only Please include all pertinent information (parent, child, divisional, or issued patent numbers) along with the appropriate serial number.

Look for any reference teach:

Text to Speech/Voice Conversion in Server, Web site or Web page ;

Subscriber terminal (i.e. client) has synthesized speech/voice generator or speech/voice synthesizer; and

the server/web site sends speech synthesizer instructions to the client terminal.

09-26-00 A09:59 IN

(see the abstract & claim 1)

FT. spe 8/27/00

STAFF USE ONLY

Searcher: Alexandra Belluck

Searcher Phone #: 308 5772

Searcher Location: PK2 4830

Date Searcher Picked Up: 9/26

Date Completed: 9/27

Searcher Prep. & Review Time: 200

Type of Search

NA Sequence (#) _____

AA Sequence (#) _____

Structure (#) _____

Bibliographic _____

Litigation _____

Fulltext _____

nt Family _____

Vendors and cost where applicable

STN _____

Dialog ✓

Questel/Orbit _____

Dr.Link _____

Lexis/Nexis _____

Sequence Systems _____

WWW/Internet _____

Other _____

Dear Examiner Fan Tsang:

Re:08/948328

Please see attached results of search Dialog databases for high quality text-to-speech which used synthesizer instruction from a network server.

Relevant references are tagged.

If you have further questions, please contact me.

Sincerely,

Aleksandr Belinskiy



Technical information specialist (SIGNAL Corp.)

EIC 2700
CPK2 4B30

Tel. 308-5172

(Search 9139 25723 9/27/00 11:40 AM)

...
...ADVANTAGE - Improves performance of **text -to-speech** system without increasing size of database used to create system

19/3,IC,K/8 (Item 1 from file: 347)
DIALOG(R) File 347:JAPIO
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06562449

EDITING SYSTEM AND METHOD USED FOR TRANSCRIPTION OF TELEPHONE MESSAGE

PUB. NO.: 20-00148182 [JP 2000148182 A]
PUBLISHED: May 26, 2000 (20000526)
INVENTOR(s): MUKUNDO PADOMANABUHAN
MICHAEL PICHENY
DAVID NAHAMUU
SALIM ROOKOSU
APPLICANT(s): INTERNATL BUSINESS MACH CORP <IBM>
APPL. NO.: 11-187372 [JP 99187372]
FILED: July 01, 1999 (19990701)
PRIORITY: 185332 [US 185332], US (United States of America), November 03, 1998 (19981103)
INTL CLASS: G10L-015/22; G06F-017/28; G10L-013/00; G10L-015/00; H04M-003/42

ABSTRACT

PROBLEM TO BE SOLVED: To correct a transcribed **text** with a **voice** by regenerating a **synthesized** speech, making a user correct the **synthesized** voice, and transmitting the corrected voice as a text through a communication system.

SOLUTION: A telephone **server** 26 transfers a text and a diagnosis to a speech **synthesizing server** 34. The speech **synthesizing server** 34 creates a **synthesized** speech and returns this **synthesized** speech to the telephone **server** 26. The telephone **server** 26 regenerates the **synthesized** speech to a user through telephone lines. One purpose of regenerating the **synthesized** speech to the user is to allow the user to correct an unacceptable or inaccurate region. The telephone **server** 26 provides the user with an option of correcting a message. The regeneration of a voice related to a correcting mechanism 36 is achieved in many methods. When the user satisfies the transcription, the telephone **server** 26 transmits the text together with a recorded voice to a message **server** 12.

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19/3,IC,K/9 (Item 2 from file: 347)
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06323595

INFORMATION DISTRIBUTION SYSTEM, INFORMATION TRANSMITTER, INFORMATION RECEIVER AND INFORMATION DISTRIBUTING METHOD

PUB. NO.: 11-265195 [JP 11265195 A]
PUBLISHED: September 28, 1999 (19990928)
INVENTOR(s): NAKATSUYAMA TAKASHI
IMAI TSUTOMU

APPLICANT(s): SONY CORP
APPL. NO.: 10-072811 [JP 9872811]
FILED: March 20, 1998 (19980320)
PRIORITY: 5538 [JP 985538], JP (Japan), January 14, 1998 (19980114)
INTL CLASS: G10L-003/00; G06F-003/16; G06F-003/16; G06F-013/00;
G06F-017/28; G10L-005/02

ABSTRACT

... SD). On the side of information receivers 6 and 7, the text information is separated from the intermediate language information and displayed out, voices are **synthesized** while using the intermediate language information, and that synthetic voice information is outputted. Namely, as the intermediate language information, **text** data for **voice synthesization** in voice **synthesizing** processing are analyzed and information made into prescribed data format is transmitted from the **server** side (information transmitters) to the terminal equipment side (information receivers).

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19/3,IC,K/10 (Item 3 from file: 347)
DIALOG(R) File 347:JAPIO
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06308270
VOICE BROWSER SYSTEM

PUB. NO.: 11-249867 [JP 11249867 A]
PUBLISHED: September 17, 1999 (19990917)
INVENTOR(s): NAMIKI IKUO
HAYASHI HIROMICHI
KANAMARU TETSUYA
KIMEDA TSUNEJI
UJIIE MASAMI
APPLICANT(s): NIPPON TELEGR & TELEPH CORP <NTT>
NTT ELECTORNICS CORP
APPL. NO.: 10-048180 [JP 9848180]
FILED: February 27, 1998 (19980227)
INTL CLASS: G06F-003/16; G06F-013/00; G06F-013/00

ABSTRACT

... BE SOLVED: To provide a voice browser system which enables even a visually handicapped person to acquire the WWW information.

SOLUTION: This system includes a **server** 100 that has a voice request acquisition means 101 which acquires a request from a client 200 via the input of voices, a voice recognition...

... which transmits a request to the URL that is designated by the client 200 based on the recognition result of the means 102 to an **internet** 70, a voice data generation means 104 which extracts a read-aloud text from the answer given from the **internet** 70 and converts the **text** into the **voice** data to **synthesize** the voices and a voice data transmission means 105 which transmits the voice data generated by the means 104 to the client 200. The system...

... which inputs the requests given from the users in voices, a request issue means 202 which extracts the URL from the result acquired from the **server** 100 and gives a request of an HTML file to the **server** 100 based on the extracted URL and a voice output means 203 which outputs the voice data received from the **server** 100.

IBM Technical disclosure/File 33 at EPOQUE

SS1 TEXT? ? (2w) (SOUND OR AUDIO? OR VOICE? OR SPEECH)
SS2 (SPEECH OR VOICE) (2N) (SYNTHES? OR GENERAT?)
SS3 S1 OR S2
SS4 (WEB OR NETWORK OR W3 OR INTERNET OR INTRANET OR SERVER? OR SITE? OR
WEB() PAGE?)
SS5 3 S 4
SS6 (SYNTHESIZ?)
SS12 /3 5 P 6
SS13 3 S 4
SS14 (PITCH? OR DURATION OR APTITUDE OR (ATTACK OR DECAY)(2N) ENVELOP? Or
(SYNTHES? () INSTRUCT?) or control? (2n) paramet?)
SS15 /2 13 S 14

? ..li

1/12 - (C) IBM CORP 1993

AN - NN9501527

TI - Techniques for Modifying Prosodic Information in a Text-to-Speech System

PUB - IBM Technical Disclosure Bulletin, January 1995, US

VOL - 38

NR - 1

PG - 527 - 528

TXT - Disclosed is a technique for modifying prosodic information in a text-to-speech synthesis system by using a sample of speech. When the generated prosody of the text-to-speech system needs to be modified, it is very difficult to teach the system correct prosody. By analyzing a sample of speech, such prosodic information as phonetic duration, pitch pattern, and stress pattern can be estimated automatically, and these prosodic parameters are used instead of the generated prosody. They are also used to retrain the prosodic models of the text-to-speech synthesis system.

Phonetic durations are estimated by using phonetic Hidden Markov Models (HMMs) for continuous speech recognition. Since the spoken text is known, the sequence of the phonetic HMMs of the spoken text is aligned with the speech sample by using the Viterbi algorithm. On the basis of the alignment, each phonetic duration is estimated. On the other hand, the pitch patterns are estimated by using a conventional pitch detector, modified to keep them within the original speaker's range. The stress patterns are also calculated from the raw power for each frame.

When these three sets of parameters of the text-to-speech synthesis system are replaced with those extracted from the speech sample, the prosody of the synthesized speech becomes very natural.

2/12 - (C) IBM CORP 1993

AN - NB9309235

TI - Voice Activated Music System

PUB - IBM Technical Disclosure Bulletin, September 1993, US

VOL - 36

NR - 9B

PG - 235 - 236

TXT - Disclosed is an approach to a voice I/O system for music or multimedia applications in which musical parameters are automatically activated based on voice command input.

- Current computer-based musical systems are based on user interaction with MIDI I/O capability (e.g., musical keyboard, PC keyboard, or music synthesizer module). This process assumes that a musician encodes all necessary musical performance information to be interpreted and processed (i.e., the burden is placed on the musician). This process is slow since all detail of the musical performance must be manually entered and cumbersome since a computer pointing device (e.g., mouse) must be used.

- The approach taken in this disclosure assumes less burden for

the musician (or end user who is not a musician) since he no longer inputs musical sequences by "hand"; rather, voice command input yield computer-generated (automated) sequences, which "fill in" or modify desired musical parameters such as pitch, chord sequence, instrumentation, style, etc.

- A System environment for a voice activated system would consist of the following. Voice Input: A voice recognition based on utterances discrete continuous (word or phrase) recognition based on utterances (i.e., lexical). Voice Output: A speech synthesizer (text-to-speech), which outputs musical parameters generated (i.e., speech output of musical parameters and not the music generated). Music Output: automated computer generated music via MIDI. Session: Musician sits in front of system, inputs via voice the musical elements desired; system outputs musical (generated) parameters and synthesized speech (output).

- The following illustrates an example session (discrete or continuous), which is in structured-English format. For discrete/continuous utterance (isolated or non-isolated words) Do:

1. Match pattern of pre-stored voice template for voice utterance.
2. Execute Matched Pattern:
If pattern found, automatically generate an output for musical attributes desired.
Else output speech error message.
3. Output as synthesized speech the musical parameters selected.

- The following pseudo-code illustrates an example how pitch would be determined.

```
/* pitch is determined as root of the chord sequence; note*/
/* N = some upper limit, and 48 = c (below middle C), 50 = D, etc.
*/
```

```
int pitch1 N =
    60, 50, 55, 48, ... ;
int pitch2 lbrc.N =
    50, 55, 48, 55, ... ;
```

```
...
int pitch1 N =
    55, 60, 48, 48, ... ;
```

```
dur=bound;
Main()
```

```
...
...
call Generate_chords(voice_input);
...
...
/* end of main */
Procedure Generate_chords(voice_input);
```

```
/* compute chordal parameters */
char string voice_input;
if (voice_input == "seventh chord first inversion")
    for (m=0;m<bound;m++)
        fputs("Chord-Note (%d,%d);\n",pitch1 m + 4,dur);
        fputs("Chord-Note (%d,%d);\n",pitch1 m + 7,dur);
        fputs("Chord-Note (%d,%d);\n",pitch1 m + 10,dur);
        fputs("Chord-Note (%d,%d);\n",pitch1 m ,dur);
        fputs("Chord-Note (%d,%d);\n",pitch1 m + 12,dur);
```

```
*rbrc. \* end of Generate_chords() */
```

The above approach is a new and novel technouque for automatic generation of music in a computer-based environment. Additionally,

current (and proposed) future music systems do not rely on voice-activated, end-user, response. Future music (enhanced) audio systems will pursue this technology, and make it pervasive across product offerings (e.g., Yamaha).

3/12 - (C) IBM CORP 1993

AN - NN92016

TI - Rule Based Speech Synthesis Method Using a Residual Codebook.

PUB - IBM Technical Disclosure Bulletin, January 1992, US

VOL - 34

NR - 8

PG - 6 - 9

TXT - - A method to synthesize natural-sounding speech for unlimited-vocabulary text by using an effectively-compressed residual source codebook is proposed here.

- BACKGROUND: In speech synthesis by rule which the LPC (Linear Predictive Coding) speech analysis/synthesis technique is applied to, the use of LPC residual signal is one of key issues to improve the quality of synthetic speech (1-4). There are two substantial problems left unsolved in applying LPC residual signal to rule-based LPC speech synthesis as follows.

(1) Quality degradation according to the pitch modification

In most rule-synthesis methods, a number of speech synthesis units (usually, several hundreds of units) are extracted from actual speech samples. To use these units for generating speech of arbitrary texts, which are different from the sample texts, the original pitch of speech synthesis units should be modified to coincide with the pitch contour of new texts. The spectral distortion caused by the pitch modification degrades the quality of synthetic speech. This type of quality degradation is more considerable in a residual-excited synthesizer than in a pulse/noise-excited synthesizer, because the residual signal is fairly sensitive to the original pitch frequency whereas the pulse signal has nothing to do with it.

(2) Sizable data of LPC residual signals for speech synthesis units

The LPC residual signal is defined as the prediction residue of LPC analysis. To use the original residual data for all the speech synthesis units causes a problem in implementing practical speech synthesis systems, such as a Digital Signal Processor (DSP)-based system which does not usually have sufficient data memory area to store the residual sources.

- This proposal focuses mainly on problem (2) above, and the result of it, conquers problem (1) in a sense. Our experimental system has about 360 speech synthesis units. The data size for spectral data (the basic part of unit data) is 80 KB. On the other hand, the residual data size is 480 KB. To improve the speech quality, many more units should be accumulated for reflecting minute contextual effects on the synthetic speech. Therefore, the problem becomes more critical for the quality improvement because the residual data size increases in proportion to the number of units.

PROPOSED METHOD: Creation of a Codebook for Voiced Residual Signals

Given this background, we propose here a method to create an effectively compressed residual source codebook without degrading the quality of synthetic speech. There are two kinds of residual signals: the voiced and the voiceless. The voiced residual signals occupy 70-80% of the whole residual data. This proposal is related only with the massive part of the residual signals, i.e., the voiced residual. As for the residual signals for voiceless speech, we use here the original signals. By the proposed method, the residual signals for voiced speech are compressed down to about one twentieth ($1/20$) without degrading the quality of synthetic speech as a residual codebook, which is created by the procedure described below.

- 1) Extraction of 1-pitch residual signal

1-pitch residual signals are extracted by observation for each frame data (4738 voiced frames in total) of all the synthesis units.

- (2) Clustering

By clustering the spectral data of residual signals using a kind of clustering methods (the LBG method (5) is used here), a residual codebook which has 256 centroids is created. The number of centroids of the residual codebook should be determined experimentally in consideration of the trade-off between the codebook size and the quality of the synthesized speech. By a preliminary experiment, we selected 256 as the codebook size.

- (3) Conversion of residual spectra to zero-phased waveforms

To use the codebook as the exciting source of an LPC synthesizer, the spectral centroids should be converted to waveforms. To compress further the waveform data without degrading the quality of synthesized speech, we adopt here the zero-phasing technique. The

zero-phased waveforms of the codebook spectra are calculated by applying inverse FFT to the spectra with the phase parts set to zero. Since the zero-phased waveform is symmetrical and its energy concentrates on the zero-point, it is very effective on the compression and robust to the pitch modification (problem (1)) in comparison with the residual signal. Moreover, the quality of the synthetic speech turns out to be fairly good and stable, mainly because the resultant codebook represents well the whole spectral space of the voiced residual signals.

- (4) Synthesis using the codebook

A residual code number is attached to each voiced frame. The zero-phased waveform centroid which corresponds to the code number is read out from the codebook and used as the exciting source of the synthesizer.

EFFECT OF THE PROPOSED METHOD: The proposed method has the following good effects, which have already been confirmed by experiments using our PC-based text-to-speech system. This method is very effective especially in the practical implementation of a high-quality text-to-speech system.

- (1) Stable quality

It is very natural that the speech quality obtained by this method is much better than that of a pulse/noise-excited synthesizer. It is, also, as good as that of a synthesizer which uses the original (not compressed) residual signals. In general, it is said that the synthesizer using the original residual has the roughness in its speech quality, because it is difficult to absorb the local fluctuation of residual signals. On the other hand, the zero-phasing

has a good effect on the overall quality which can make it homogeneous and stable because of the robustness in the pitch modification. Moreover, since the codebook represents well the whole spectral space of the voiced residual signals, the homogeneous quality is not so far from that of a synthesizer using the original residual signals. These are the reasons why the speech quality of the proposed synthesizer is as good as that of a synthesizer using the original residual signals.

(2) High data compression rate

Not only the stable quality but also a high data compression rate can be obtained by this method. For instance, 1/20 is the compression rate of our experimental system.

The original residual data size;

4738 (frames) x 80 (points) x 1 (byte) = 379.04 (KB)

4738 (frames): number of voiced frames

80 (points): average number of points per 1-pitch residual signal

1 (byte) : data size per point

The data size compressed by this method;

256 (centroids) x 64 (points) x 1 (byte) = 16.384 (KB)

256 (frames): number of centroids

64 (points): number of points per 1/4-pitch zero-phased residual signal

1 (byte) : data size per point

The compression rate;

(16.384 / 379.04) x 100 = 4.3 (%) => less than 1/20

References

(1) Arai, "Experiments on the Exciting Sources for LPC Speech Synthesizer," Journal of the Institute of Electronics, Information and Communication Engineers, J69-A,12, pp.1555-1563, 1986. (In Japanese)

(2) Hamada, "Study on Exciting Sources for Speech Synthesis by Rule Considering the Residual Spectrum," Proc . of the Autumn Meeting of the Acoustic Society of Japan, 1-1-6, 1987. (In Japanese)

(3) Hirokawa et al., "Exciting Sources for Rule-Based Speech Synthesizer using Residual Signals," Proc . of the Autumn Meeting of the Acoustic Society of Japan, 2-2-11, 1986. (In Japanese)

(4) Iwata et al., "A Rule-Based Residual-Excited Speech Synthesizer," Proc . of the Autumn Meeting of the Acoustic Society of Japan, 3-2-7, 1988. (In Japanese)

(5) Y. Linde et al., "An Algorithm for Vector Quantizer Design," IEEE Trans . COM-27, pp.84-95,1980.

4/12 - (C) IBM CORP 1993

AN - NN9111206

TI - Flexible High-Quality Audio Delivery Via Infrared Link.

PUB - IBM Technical Disclosure Bulletin, November 1991, US

VOL - 34

NR - 6

PG - 206 - 208

TXT - - Disclosed are flexible, modular approaches to providing high-quality stereo audio to the personal computer or workstation user. Included is a description of a novel PC-speaker in a chair implementation.

- As multimedia systems become more prevalent, full-range audio signals will replace simple "beeps" as audio output from personal computer systems. The traditional audio speaker location within the computer system unit will quickly be found inappropriate for delivery of this type of audio. Adding larger, higher-quality speakers to the computer system is one approach; however, the volume required to

- assure that adequate sound reaches the user will cause a nuisance factor in the home or office environment. Obviously, alternate methods for delivering high-quality audio to the computer user are needed.
- Depending on the environment in which the computer system is used, various approaches to delivering this sound may be appropriate. For example, high-fidelity amplified speakers would be appropriate where the computer is being used to present information to several people in a meeting room, while use of such speakers in a typical office environment would be intrusive, and headphones might be more appropriate. Because the use environment for a particular system cannot be predicted in advance, a flexible approach to delivering audio to the user is required.
- The approach disclosed here allows the use of a multitude of delivery systems, including those mentioned above, all of which receive audio information modulated by an infrared (IR) signal. Versions of this approach may be built into a computer system or added to an existing system.
- The basic approach used in all cases disclosed here is one in which the audio signal from the host computer is modulated onto an IR carrier. The IR sending module is located in a convenient place, such as atop the keyboard or CRT. A variety of devices receive the IR signal and demodulate it back to audio.
- Two implementations of the sender unit are disclosed. In the first case, it is assumed that the host system has been designed with existing audio-out jacks. Here a standalone modulator unit is used. A plug may be inserted into the audio source jack of the host computer. A cable brings the audio information into the unit body. Within the body is modulating circuitry and an IR output diode. A lens diffuses the IR output omni-directionally. The body may be mounted on top of a CRT, keyboard, or system unit, as appropriate for the particular system set-up. Adapter plugs are provided to allow simple adaptation to many jack styles, including mini-phone, phone, and RCA cables. The circuitry to implement such a sending unit is commercially available. The circuitry may be powered by a battery or AC adapter, or provisions may be made to allow a DC voltage to be provided by the host computer and transmitted via the connecting cable.
- The function and circuitry for the second embodiment of the sender unit is identical, with the difference being that the entire system is integrated into the computer system, rather than being an add-on feature. This approach has certain advantages, as the cables can be integrated into existing keyboard or CRT cables, and the diode lens can be integrated into the hardware in an attractive way.
- Different receiving unit types are proposed. In each case, the receiving unit contains an IR detector and demodulation circuit. Again, this circuitry is commercially available. Each type of receiving unit is described briefly in the following. In one embodiment, amplified speakers are used which snap onto the right and left side of the keyboard. The receiving circuitry drives a low-power amplifier contained within small speaker units. The speaker units are designed to match the cosmetics of the keyboard and other personal computer hardware, and contain full-range speakers up to 5 inches in diameter. Speakers this size can typically reproduce audio in the range of 100 Hz - 12 KHz. This range covers the vast majority of human hearing. Since the user is, by definition, within arm's length of the keyboard and speakers, low audio levels can be used to provide sufficient volume to the user, while interference and distraction to others is minimized. A peg-and-hole arrangement is one of many physical design techniques which could be employed to

allow the speakers to be physically locked to the keyboard. Alternatively, the speakers may be placed in any convenient location in the vicinity of the keyboard. A related embodiment uses speakers separated by long wires from the keyboard.

- In another embodiment which involves a PA/audio system input, a unit is proposed which simply translates the received IR signal back into an audio line-level signal and outputs it to RCA plugs, which may be used to connect to an existing PA or Hi-Fi audio system. This approach may be most appropriate for home use, as it minimizes the investment by using equipment already in place. Additionally, this is a useful approach for auditoriums or when the host computer is being used to drive an audio/visual presentation to a large group.
- In a chair/integrated speakers embodiment, an ergonomically designed high-back chair contains small high-quality audio speakers mounted inside the chair back, one just behind each of the user's ears. Again, the audio signal is transmitted via infrared link from the host PC to the chair. The chair body contains circuitry which receives and amplifies the signal, and presents it to the user via the speakers. A volume control may be located in any convenient spot, perhaps hidden in the armrest. The top sections of the chair back adjust to fine-tune the location of the speakers to fit the individual user. The chair is a novel approach to delivering high-fidelity sound to the computer user. Because of the close proximity of the speakers to the user, very low volume levels are sufficient, and so there is little possibility of distraction to those nearby. However, the disadvantages of headphones (fatigue, lack of comfort, inability to use the phone, easily lost, can shut out external sounds, sanitation) are avoided. Note that the headphone-like quality of the speaker chair allows use of binaural sound sources and other psychoacoustical effects for three-dimensional or surround-sound effects without other special equipment. This type of chair would be especially useful in productivity centers, or multi-media learning labs where many users may be working in close proximity. In related embodiments, the speakers may be snapped on to existing chairs in a user's office, and a circuit can be used which turns off the audio when no one is sitting in the chair.
- Standard headphones with IR receiving circuitry could also be used to receive high-quality audio from PCs.
- Note the fact that the delivery of remote audio may have particular value in areas where the computer system unit (and integrated speaker) are inaccessible or behind a wall, for example, in museum displays, shopping malls, and public information centers. In addition, the remote wireless delivery of audio information may be useful in schools. The ideas disclosed need not replace the traditional speaker; in fact, they can be used in conjunction with a system speaker in the following way. Audio intended for a single user can still be broadcast over the local computer system-unit speaker, while general information or emergency information can be transmitted from the computer to a remote audio delivery system, as described. This invention also applies to the remote wireless delivery of other information relating to audio, such as MIDI signals for musical instruments, and speech synthesizer control parameters.

NR - 6B

PG - 390 - 391

- TXT - - This article describes a method for controlling pause duration in spoken sentences synthesized by a text-to-speech system. This method is based on analysis of spoken sentences and can produce natural pauses.
- In Japanese, pause duration is very important for communicating the syntactic and semantic structure of the sentence. Consequently pause duration control is a key to synthesizing natural-sounding spoken sentences.
 - Conventional methods: There are two typical methods for controlling pause duration. The first method is based solely on punctuation marks; pause duration P is given by:
$$P = P_0 \text{ (No punctuation mark)}$$
$$P = P_1 \text{ (Punctuation mark),}$$
where P0 and P1 are constant values of pause duration.
***** SEE ORIGINAL DOCUMENT *****
The other method controls pause duration solely by using breath-group length before the pause (L1); pause duration P is given by:
$$P = a' + b' * L_1,$$
where a' and b' are parameters given by regression analysis.
 - However, pause durations generated by using these methods bear little relation to those of natural utterance data. Pause durations assigned by these rules do not contribute to the naturalness of the synthesized speech.
 - New method: In this new control method, pause duration is calculated by the length of both the breath-groups before and after the pause (L1, L2); pause duration P is given by $P = a'' + b''(L_1 + c'' * L_2)$, where a'', b'' and c'' are parameters given by regression analysis.
 - Fig. 2 shows the relation between pause duration and the length of the breath-groups both before and after the pause. Pause duration data are dotted around a straight line, and the correlation of the regression line (R) is very high.
 - This new method of controlling pause duration contributes to the naturalness and intelligibility of the synthesized speech.

6/12 - (C) IBM CORP 1993

AN - NN8710208

TI - Mechanism for Integrating VOICE and DATA on a Transmission Channel

PUB - IBM Technical Disclosure Bulletin, October 1987, US

VOL - 30

NR - 5

PG - 208 - 209

- TXT - - A way to integrate voice and data information on the same transmission channel consists in reducing the rate needed for voice transport by means of voice-compression techniques. These techniques are sophisticated, and voice-compression equipment is needed at both ends of the channel. This article relates to a simple mechanism allowing voice and data to be transmitted on the same channel without using voice-compression techniques. Even if no compression technique is used, voice signal can be carried together with data information, due to the fact that voice signal includes no-activity periods. Such periods correspond to a voice level lower than a predetermined

threshold. Such no-activity periods are long compared to a slot duration which is generally of 125 microseconds. A voice activity detector VAD detects inactivity with an integration of the voice signal over several slots. In this environment a difficulty arises to delimitate the voice and data information since non-compressed voice slots must not be permanently altered and it is not possible to use one bit out of the n bits in the slot to indicate whether this slot carries voice or data. The main idea is to use an HDLC (High Level Data Link Control) flag F as a delimiter, and a slot handler insures that the voice slots never simulate a flag. It is to be noted that the zero-insertion techniques cannot be used inside the voice stream, as it is necessary to keep all voice slots at slot boundaries. The slot handler detects voice slot values corresponding to flags F, and alters them to avoid flag simulations by changing the flag pattern 01111110 into 01111111; the low-order bit is changed from 0 to 1. This does not cause any significant degradation of the voice quality. Once the flag simulations have been eliminated from the voice information, it is possible to use flag F to indicate the beginning and the end of a no-voice activity period of time that will be used to carry data information on the same channel.

/ DATA / F / VOICE / ----- / VOICE / F
 ---- ---

No voice activity The zero-insertion/deletion technique applies to the data stream, to avoid false flag simulations during data periods. If data stream corresponds to an HDLC transfer, zero insertion applies to all data, including the message flags. The voice activity detector VAD detects the no-voice activity periods during which data can be transmitted and handled in a conventional way by means of zero-insertion circuit and flag generator. The merged voice and data stream is transmitted on channel CH. At the end of a voice activity period, a flag is generated at a voice slot boundary to indicate that next bytes are voice bytes. This means that the data portion is a multiple of the voice slot duration, but corresponds to any number of data bits due to the zero insertion. If a zero is to be inserted when a flag must be generated, the first bit of the flag is considered as the inserted zero. Consequently, the voice slots do not suffer any delay distortion; they are delivered as if they used the whole channel for themselves. This allows the receiving end to use a normal decoding circuit. Data portions correspond to voice idle periods which are distinguished at the receiving end as the period delimited by flags during which the receiver generates a permanent idle signal.

7/12 - (C) IBM CORP 1993

AN - NN86123055

TI - Constructing Method for Speech Synthesis Units

PUB - IBM Technical Disclosure Bulletin, December 1986, US

VOL - 29

NR - 7

PG - 3055 - 3057

TXT - - A segmentation and smoothing method is proposed to build smoothly connectable speech synthesis units from human utterances.

Background Diphone, as a speech synthesis unit *, enables smooth

connection and sophisticated duration control. However, it is difficult to build a diphone which works in various phonetic environments. Some phonemes are strongly co-articulated or need allophones to keep intelligibility and naturalness. Also, in combining synthesis units to synthesize a word, sentence or text, smoothing is required to avoid a perceptual discontinuity between connected frames caused by changeable vocal effort. VCV (Vowel Consonant Vowel) - based diphone In this proposal, diphones are adapted to include co-articulations or allophonic features by additional entries for specific phonetic ***** SEE ORIGINAL DOCUMENT ***** environment. In a mora-based phonetic system such as Japanese, these problems are solved by extracting parameters from VCV segments without losing freedom of duration control. In a VCV-based diphone set, only a pair of (V1C) and (CV2) diphones from the same V1CV2 segment can be connected with each other at the consonant portion. Note that, for example, of 5 Japanese vowels /a,e,i,o,u/ and consonant /r/, 5 different kinds of (ar) diphone must be prepared for each succeeding vowel, and 5 kinds of (ra) diphone must be prepared for each preceding vowel. Fig. 1 shows the example of proposed segmentation. In Fig. 1, points a, b, and c are determined by spectral features and signal power. When a consonant is continuant, redundant frames around point b are omitted. Normalization and smoothing of parameters To eliminate perceptual discontinuity, synthesis parameters, such as amplitude and formant frequencies, should be identical to those of neighboring diphones at the connecting point. Proposed here is a simple method to smooth synthesis parameters. Series of raw parameter values extracted from human speech are: 1) normalized to a unique value which is determined previously at the vowel end-frame, and 2) smoothed according to linear interpolation at the other frames. Smoothing is performed within VCV, and then it is split into two diphones to prevent modifying transition unnecessarily. Fig. 2 shows the process of smoothing. In Fig. 2, one of formant frequencies is smoothed, which is identical to "normal value" fv1 and fv2 at both end-frames, respectively, and can be connected to the preceding (-V1) diphone and the succeeding (V2-) diphone without discontinuity. Reference N. R. Dixon and H. D. Maxey, "Terminal Analog Synthesis of Continuous Speech Using the Diphone Method of Segment Assembly," Trans . IEEE, AV-16, 40-50 (March, 1968).

8/12 - (C) IBM CORP 1993

AN - NN86055462

TI - Generation of Nasalized Vowels in Text-To-Speech Synthesis

PUB - IBM Technical Disclosure Bulletin, May 1986, US

VOL - 28

NR - 12

PG - 5462 - 5463

TXT - - The present method involves synthesizing the nasalization of vowels between consonants in a speech synthesis environment.

Briefly, (a) primary speech units --such as diphones-- which are concatenated to form words are scanned for the presence of a nasal consonant, (b) a look-ahead is performed to detect the presence of a second nasal consonant, and (c) if a second nasal consonant is detected, a nasal branch of the synthesizer is turned on for the

duration of the intervening vowel. In describing the method in further detail, it is observed that for most phoneme or diphone formant synthesizers, there are 10 to 40 control parameters guiding the synthesizer in producing a speech waveform. These parameters change through time; the entire time ensemble for each parameter-class is often referred to as a "channel". One common parameter is called AN (amplitude of nasality). By way of an example, let AN_i be the amplitude of nasalization as a function of time for a synthesized speech utterance. $AN = 0$ would imply no nasalization. First the detection of the presence of steady-state nasalization at a particular time point, i , must take place to trigger the algorithm: $AN_i > 0$ and $AN_{i+1} = AN_i$ $i = 1, 2, 3, \dots$

1 If the above condition (Eq. 1) is true for a particular i , then i is saved, and a search is conducted for a future region of steady-state nasalization from $i + t_1$ to $i + t_2$ (for example, $t_1 = 5$ ms and $t_2 = 30$ ms). Too long a future search (large t_2) would lead to unwarranted nasalization. The search may be easily performed by searching the future AN's for a value equal to the current detected steady-state value at i : $AN_j = AN_i$ $j = (i + t_1), (i + t_1) + 1, \dots, i + t_2$

2 t_1 is needed to preclude the current nasal consonant from the search. If Eq. 2 is true for a particular j , then the intervening vowel sound is nasalized by turning the nasal synthesizer branch on up to time point j : $AN_k = AN_i$ $k = i, i + 1, i + 2, \dots, j$

3 Eq. 3 is only implemented if there is no time between the surrounding nasal consonants during which voicing is interrupted (i.e., $A_0 \neq 0$, where A_0 is the amplitude of voicing). Since nasalized vowels are constructed algorithmically by this approach, it is not necessary to store diphones containing these sounds, and the size of the library of stored sounds is not increased as a result. The above-outlined method is illustrated by the following example: Input example text: "man" Phonemic transcription: MX AE NX Diphone transcription: MXAE AENX Result generated from algorithm of method: MXAEn AEnNX (n indicates nasalization).

9/12 - (C) IBM CORP 1993

AN - NN86055427

TI - Generation of "H" Sounds in Text-To-Speech Synthesis

PUB - IBM Technical Disclosure Bulletin, May 1986, US

VOL - 28

NR - 12

PG - 5427 - 5428

TXT - The present invention relates to a method for producing high-quality "H" sounds in a speech synthesizer. Because many speech synthesis systems construct utterances from a database of stored steady-state sounds (phonemes), or transitions between steady-states (diphones), it is necessary to have a steady-state description for each sound. However, the /h/ sound is so influenced by the characteristics of its surrounding sounds that it cannot be defined and stored as a steady-state phonemic unit on its own. Similarly, in the case of diphone synthesis, this chameleon effect makes it impossible to define transitions to a generic steady-state "H". A method for producing high-quality "H" sounds using diphones as primary units is now described, and the same underlying principle could be applied to phoneme synthesis as well. In brief, the input string of diphones is scanned for the presence of the "H" sound. When found, the proceeding sound is tapered to silence, a transition state is constructed from an already existing unit, and the following sound is started with a gradual onset from silence. The method can be

defined more rigorously and more generally in terms of the following diphone notation. Each diphone is represented as a pair-transition $p(n):p(n+1)$, $n = 1, 3, 5, \dots$. The string of diphones making up an utterance is scanned until $p(n+1) = "HX"$, at which point new diphones are inserted. By way of example, suppose there are two pair-transitions characterized as: $p(1):p(2) = EEHX$ and $p(3):p(4) = HXEH$. Given the detected diphones containing the H sounds, ***** SEE ORIGINAL DOCUMENT ***** where $p(2) = "HX"$, the following transformation from two diphones to three diphones is applied: ***** SEE ORIGINAL DOCUMENT ***** XX indicates silence, and $p(n):XX$ therefore indicates gradual tapering of the $p(n)$ sound to silence. "Ah", "asp", and "A0" are typical control-data parameters (and notation) for speech synthesizers. "Ah" is the amplitude of the hiss source (random number driving function). "Asp" is a bit that indicates aspiration (the noise source directed through the formant chain). "A0" is the amplitude of voicing. In other words, a transition $p(1):p(4)$ is constructed using a pre-existing diphone with subsequent modification (application of a low level of aspiration during the smooth transition to obtain a natural "H" sound). During $p(1):p(4)$ any voicing or nasalization in the original diphone is turned off ($A0=0$, $A_n=0$). Since all "H"-sounds are constructed algorithmically by this approach, it is not necessary to store diphones containing these sounds, and the size of the library of stored sounds is therefore decreased.

10/12 - (C) IBM CORP 1993

AN - NN85081248

TI - Use of the Grid Search Technique for Improving Synthetic Speech Control-Data

PUB - IBM Technical Disclosure Bulletin, August 1985, US

VOL - 28

NR - 3

PG - 1248 - 1249

TXT - - Many speech synthesizers utilize a library of stored control-data parameters to direct the actual software synthesizer in producing the output speech waveform. The number of such parameters varies with the type of synthesizer, but usually is within the range of 10 to 40 parameters. The method described here would be useful in optimizing the values of such parameters so that the synthetic speech power spectrum (amplitude vs. frequency) most nearly conforms to natural speech power spectra. Traditionally, the grid search technique is used to fit curves with simple mathematical expressions (such as gaussian, trigonometric, or polynomial functions) to experimental data. Here, the technique is applied to a mathematically complicated function, the synthetic speech power spectrum, which cannot be described by a simple algebraic expression. The Method Let a measure of goodness of fit X^2 between the synthesizer power spectrum S_i and natural target spectrum N_i be defined as: ***** SEE ORIGINAL DOCUMENT ***** where s , the uncertainties in the natural spectral points, may be set to 1 for this discussion. The synthesizer spectrum is a function of the control-data parameters c_j . X^2 may be considered a continuous function of the parameters c_j describing a hypersurface in n -dimensional space. The space must be searched for the appropriate minimum value of X^2 . The optimum values for c_j can be estimated by minimizing X^2 with respect to c_j . Step 1) Initial values for c_j

are given by the current control-data parameters. Step 2) One parameter c_j is incremented by a quantity W_c (user-selected), where the program chooses the sign such that X_2 decreases. Step 3) The parameter c is repeatedly incremented by W_c until X_2 starts to increase, and the minimum value is determined by parabolic interpolation. Step 4) X_2 is minimized for each parameter. Step 5) The above procedure is repeated until the last iteration yields a negligibly small decrease in X_2 . Applications The current synthesizer control-data c_j serves as input to the grid search. The final values of c returned by the algorithm direct the synthesizer to produce a power spectrum most nearly like the human speech spectrum, and these new c values may be stored in the library in place of the old values. Since the mathematical similarity between natural and synthetic speech curves may not necessarily correspond to perceptual similarity, sets of parameters may be saved near the minimum X_2 for subsequent perceptual testing. The 'best' parameters may then replace the old values within the library. The method outlined is itself computationally fast and has a minimum number of assumptions as prerequisite for its use. The power spectra may be smoothed prior to the grid search in order to eliminate pitch as a variable in the calculation. This technique can provide an aid to achieving the goal of almost all speech synthesis: the production of a natural and intelligible speech output.

11/12 - (C) IBM CORP 1993

AN - NN83113071

TI - General-Usage Remote-Access Storage and Forward Message Handling

PUB - IBM Technical Disclosure Bulletin, November 1983, US

VOL - 26

NR - 6

PG - 3071

TXT - - The technique discussed herein enhances the ability of a telephone desk set to offer automatic call answering and store and forward capability. The logic processing discussed can be applied to any telephone or private branch exchange (PBX) system. The General-Usage Remote-Access Storage and Forward Message Handling allows a caller to leave information at either a busy or unattended telephone. The telephony system being discussed allows for acquisition of message information without recourse to voice digitalization/storage. The telephony management system uses speech synthesis to advise a caller that the phone is either unattended or busy. A canned, synthesized message is used for these purposes. The caller is advised that by using his push-button key pad he can leave his telephone number by simply rekeying it in. The caller is then prompted to leave his name by the following push-button sequence for each character in the caller's name: 1. The push-button key containing a respective character of the caller's name is touched. 2. Immediately afterwards, the number 1, 2, or 3 is touched to indicate which character on the previous stroked key was the intended entry. In this manner, a person's name can be spelled. Q and Z are entered as if they were inscribed on the "1" push button. A priority can also be entered by striking the appropriate push buttons as prompted by the speech synthesized instructions. Hence, "1" can indicate urgent, "2" return this call at your convenience, and "3" return this call today. The system can then automatically time stamp the call. All instructions and prompts that the caller hears are

speech synthesized. This allows precise, clear instructions to be canned when the system is produced yet leaves the telephone owner the prerogative of adding a personalized introductory or ending message. This is done by composing the personalized message in machine-readable form and then having it synthesized and appended to the canned message. Having the appended message enunciated by the synthesizer avoids introducing another voice into the message a caller hears. Messages are gotten from the phone either via a CRT or by a canned voice on the phone spelling the caller's name, giving the telephone number and priority.

12/12 - (C) IBM CORP 1993

AN - NB80123478

TI - Audio Indication of Error in Speech Recognition. December 1980.

PUB - IBM Technical Disclosure Bulletin, December 1980, US

VOL - 23

NR - 7B

PG - 3478 - 3479

TXT - 2p. A technique is described for providing an audio indication of the recognition reliability in speech recognition and altering the speech quality in the speech synthesizer.

- In a recently proposed speech compression technique, a speech signal is recognized by means of a speech recognizer and thereby converted into a string of words or phones (units of vocal sound). The resulting string of words or phones is then transmitted to a distant location where the speech signal is resynthesized. The problem that arises in such a compression technique is that when the speech recognizer makes an error, an incorrect word or phone is synthesized which sounds as good to the listener as the correctly recognized words or phones.
- The subject disclosure provides an indication of the speech recognizer's reliability as an auxiliary signal which is transmitted in addition to the word or phone string. A speech recognizer 1 uses a reliability estimator 2 to estimate its own reliability from the likelihood profile for the word or phone in question or from some other suitable measure. The reliability indicator is used by a speech synthesizer 3 to alter the quality of the resynthesized speech. Words or phones with high reliability are resynthesized with little alteration, while words or phones with low reliability are modified during resynthesis.
- One method of modification is to add noise via a generator 4 to the synthesized speech, or to the control parameters of the synthesizer. Another method is to transmit, in addition to the reliability estimate, an alternative word or phone string which is resynthesized and mixed with the primary speech, in proportion to the reliability estimate.
- In this manner, speech transmitted by recognition-transmission-synthesis is provided with an indication of its reliability. The reliability is indicated orally, permitting the listener to use his own well-developed auditory sense in an attempt to reconstruct the correct signal.

fo ss 12

? ..fo ss 10

1/13 - (C) IBM CORP 1993

AN - NNRD426114

TXT -

1.4

... first two points above relate to converting back-end data from its server-dependent format to the infrastructure's canonical representation. When pulling data items from back-end sources, the...

1.8

For example, all IMAP4 mail servers are serviced by our IMAP4 facade. Similarly, all common news servers can be serviced by an NNTP facade. So, in practice, O...

1.8

...ther reducing the number of number of facades, we designed a facade for a point web source, which is simply the contents of a single URL. Common examples of a point web source include stock quotes and weather forecasts. Adding another such source simply ...

1.12

...lication programmer will likely implement a new program to do the formatting; an experienced web publisher will likely choose the JS...

1.13

...M> into <SENT>,

? ..li

1/13 - (C) IBM CORP 1993

AN - NNRD426114

TI - Multi-modal Data Access

PUB - IBM technical Disclosure Bulletin, October 1999, UK

NR - 426

PG - 1393

TXT - With the proliferation of pervasive devices such as cellular phones, smart phones, Palm Pilots (WorkPads) and other PDAs, it is becoming necessary to provide multi-modal access to personal data such as e-mail, calendar and address book. And, as one would expect, such solutions are emerging.

However, most (or all) of these solutions tightly integrate the user devices to the back-end. For example, one company might provide access to e-mail through voice. Another might offer calendar through browsers and PalmPilots.

In this paper, we describe an open, standards-based approach to this problem. Rather than specifying which back-ends are accessible, we define an open method for adding back-ends. Similarly, we describe an easily extensible mechanism for producing clients implementing various modalities.

In designing our solution, we assumed that we were not permitted to alter either the data sources or the devices. Thus, we must access the sources using whatever protocols they currently export, and we must deliver the content to the devices in whatever format(s) they can render.

Our resulting infrastructure consists of three layers: interfaces to back-end data sources, which we call facades; an input processor, which we call a request multiplexer, or ReqMux; and a set of output formatters. Below, we describe each of these components. As shown in Figure 1, each request flows from a client into the ReqMux, which passes the request to a facade, which passes the results to an output formatter. A request contains a request type, such as get

today's calendar entries or get message N, and a client-device indicator, which is used to determine the output format, as described further below.

Facades are the liaisons between our infrastructure and the data sources. Each facade has three basic responsibilities: extract data from a source; convert the data from the source-dependent format to our canonical representation; and export the sources's commands to the remainder of our infrastructure.

The first two points above relate to converting back-end data from its server-dependent format to the infrastructure's canonical representation. When pulling data items from back-end sources, the facade is we assume that the sources are fixed -- that is, they will not be modified to accomodate our infrastructure -- so any

changes in data format required by the infrastructure must be made by the facade. Since the source is fixed, each facade must use the protocol exported from the source. For example, we've implemented POP3 and IMAP4 facades for mail retrieval.

Once the facade has extracted data from a source, it transforms the data into a canonical internal format. This representation allows the system to normalize differences among sources. We use XML for our representations. For example, the mail facade can produce an XML document representing an inbox, such as:

```
<?xml version="1.0" encoding="US-ASCII" standalone="no"?>
<!DOCTYPE Mail SYSTEM "MML.DTD">
<Mail>
  <MessageSummary id="m1">
    <Date>Fri Jan 29 13:58:36 EST 1999</Date>
    <From>rak@us.ibm.com</From>
    <Subject>Kenny</Subject>
  </MessageSummary>
  <MessageSummary id="m2">
    <Date>Fri Jan 29 13:59:02 EST 1999</Date>
    <From>dlk@us.ibm.com</From>
    <Subject>Cartman</Subject>
  </MessageSummary>
</Mail>
```

To accomodate the various client devices, this XML representation is reformatted by the output formatter, as described below.

Facades also define the commands that are valid for their respective sources. When facades register themselves with the ReqMux (described below), they pass a handle to themselves, along with a list of supported commands. For example, a mail facade might support get inbox, get message N and delete.

As we describe further below, interpreting certain of these commands requires knowledge about the state of source, and this state information is stored at the facade. For example, the e-mail command get next only has meaning if one retains the index of the last message accessed.

Finally, to improve performance, facades can cache data from their sources. Like all caching, this should be transparent both to the source and to the remainder of the infrastructure.

Since facades typically only implement a single protocol, each time a new type of source is added, a new facade must be written. Fortunately, for many common cases, standard protocols exist, so facades can be reused. For example, all IMAP4 mail servers are serviced by our IMAP4 facade. Similarly, all common news servers can be serviced by an NNTP facade. So, in practice, only

a small number of facades are required.

Further reducing the number of number of facades, we designed a

facade for a point web source, which is simply the contents of a single URL. Common examples of a point web source include stock quotes and weather forecasts. Adding another such source simply requires adding a new URL and corresponding command (e.g., <http://www.weather.com, get weather>) to the point source facade. This can be accomplished through an HTTP request sent to the point source facade.

The ReqMux receives client requests, and passes them to the facades for processing. Its primary function is deciding which facade should handle an incoming request. The ReqMux maintains a list of <command, facade> pairs. When it receives an incoming request, the ReqMux uses this list to determine which source will handle the request.

In some cases, there is a single source for a request, and the choice is straightforward. For example, if the request is get inbox, then that will be routed to the mail source.

However, some requests can be handled by multiple sources, with the proper choice determined by the system state. The ReqMux maintains enough state data to handle such cases. For example, the get next request might be valid for both a mail source (get the next message) or a calendar source (get the next meeting

on the calendar). In this case, the request is routed to the source that handled the previous request. If no request preceded this one, or if the source for the previous request does not support the ambiguous request, then an error is triggered.

Once the data are retrieved from a source and converted to canonical form, they are ready to be formatted for the client device. Recall that a client-device indicator flowed in the initial request into the infrastructure, and it is preserved as data flow from component to component. That indicator is used to select an output formatter appropriate to the device.

Different devices can require different modalities (e.g., speech vs. HTML) or they might impose different constraints within the same modality (e.g., a PC browser vs. a PDA browser). Each supported variant requires an appropriate formatter.

We considered three ways to implement the formatter: Application Program; XSL based Style Sheet script; and JSP based script.

These choices vary in how they describe the transformations needed. The transformation implementor will likely choose among the technologies based on personal preference and expertise. An application programmer will likely implement a new program to do the formatting; an experienced web publisher will likely choose the JSP

based approach; and SGML authors will likely favor the XSL approach.

In our prototype system, we include two output formatters, one for speech, and one for browsers. The speech formatters transforms the canonical format into JSML; the browser formatter transforms it into HTML suitable for both PCs and PDAs. We implemented both of those formatters using the Application Program technique, as well as the XSL style sheet technique.

We considered two ways to exploit the application program technique. The first manipulates the in-memory DOM tree; the second leaves the DOM tree intact, but changes the way it is printed. (Recall that a DOM tree in an in-memory representation of an XML document.) Both variants begin by using standard DOM APIs to read the XML document, and produce a corresponding DOM tree. We use IBM's XML4J parser to create the DOM tree.

When manipulating the DOM tree in memory, the goal is to find the nodes of the tree that contain the tag to be replaced, and to change

the text in those nodes into the new tag text. For example, our mail messages contain a <FROM> tag, but that tag has no meaning in JSML. We choose to translate <FROM> into <SENT>, which is the JSML sentence tag, and causes the voice synthesizer

to read the entry as a sentence. (Note that both the <FROM> and tags are represented by the same node of the DOM tree. Consequently, changes to the "FROM" text affects both delimiters.)

In XML parlance, tags are represented in the DOM tree by nodes of type Element. Thus, the algorithm is to examine the entire DOM tree searching for Elements, and when an Element is found, to compare the Element's text to the target text. In our example, we're looking for FROM. When we find a match, we use an XML4J

method to change the name to the new text, in our example, this changes FROM to SENT. (Note that this method is not part of the DOM standard; it was added by the XML4J developers.) When this completes, all FROMs are SENTs.

There is one further complication: we don't simply want to speak the text delimited by the <FROM> tag; we also want to speak the word "from." This requires that we insert a text node into the DOM tree as the first child of the SENT element. This text node contains the word "from." This causes the synthesizer to speak

"from" before speaking the text in the <FROM> field. We insert this node using another XML4J method.

When manipulating the string representation of a DOM tree, instead of changing the internal representation of the tree, we change the way the DOM tree is rendered as a string. To convert the tree to a string that embodies the formatted XML document (either JSML or HTML), we execute code to traverse

the tree, rendering the DOM tree node-by-node.

To convert the DOM tree to a string, we must visit each node in the tree. Conveniently, XML4J comes with several classes that automatically visit each node in a DOM tree. They vary in the order in which the nodes are visited. We use the NonRecursivePreorderTraversal class.

This class takes as a parameter a class that implements the Visitor interface, where Visitor refers to the design pattern of that name. (1) The Visitor interface is used to perform operations on each node of a DOM tree, and the operation performed depends on the node's type.

The Visitor interface requires that methods be defined for each DOM-tree node type. However, since we are only altering tags (in our example, "FROM" tags), which are Element nodes in the DOM tree, all other types of node are left unchanged. Thus, in our subclass that implements the Visitor interface, for all other

types, the methods do nothing. In our Element-handling method, we compare the Element's text to the target text. If the text does not match the target, the text is printed; if it does match, the replacement text is printed.

In summary, both techniques described above are quite similar: they both examine each node in the DOM tree, searching for test of node type (that is, the test of whether the node is an Element) is made explicitly by the programmer's code; when using the XML4J's Visitor pattern, the base class does the test for us, and simply calls an appropriate method when an Element is encountered. Also, the Visitor technique leaves the DOM tree intact, which permits us to perform additional operations on the original form. After considering both techniques, we chose to implement the Visitor technique.

The XSL style sheets express through a pattern matching language

what transformations are to be performed. The style-sheets are applied by means of chaining the data source servlets with a servlet developed by our team in conjunction with the WebSphere Application Server (WAS) team. The XSL style-sheets we developed are stored in the WebSphere, and selected based on the output type requested by the client.

Other formatters based on JSP technology are conceived but remain unimplemented.

The choice of formatting technology will vary widely. We expect the choice to be primarily based on the knowledge and experience base of the implementor rather than on the goodness of any particular technology.

Where possible, our prototype leverages existing web infrastructure. As shown in Figure 2, each component listed above is implemented as a servlet written in Java. Parameters passed from clients to the infrastructure (e.g., device indicators) are embedded in HTTP requests. Our servlets are tied together via servlet chaining as implemented by IBM's WebSphere product. We use Apache as our HTTP server.

The ReqMux must know about the available sources. In our first prototype, this information was configured statically. A systems administrator configured the ReqMux with a list of available sources and the commands available for each source.

We later augmented our ReqMux and facades to allow dynamic registration. When a facade comes on-line (with its corresponding source), it registers with the ReqMux via an HTTP flow. The facade passes its URL and the list of valid commands.

The ReqMux must then update its request-routing table to reflect the new sources. Commands that do not overlap with any registered commands are simply added to the table. If the same command has previously been registered by another facade, then the conflict must be noted in the table. Invocations of such requests are resolved using state information, as described above.

This system allows users to access multiple data sources seamlessly. However, access to multiple data sources requires that the user be authenticated to each source. One could require that the user enter the password for each source individually, but this is overly cumbersome.

Commercial global sign on (GSO) products handle this problem by allowing the user to enter the passwords for each data source once. The user authenticates once with the infrastructure, which then acts on his behalf when interacting with the sources.

We created a modestly secure GSO system for our prototype. Our system is clearly not sufficiently secure for commercial deployment, but it served our purposes.

The point of this system is to allow users to access sources multimodally. In the easiest case, they use a browser to generate the HTTP requests that drive our system. Typically, the process begins when a user enters the system login URL, and is challenged for a user ID and password.

If the user successfully authenticates, the system returns its main web page. This page includes the list of valid requests, and the user selects by clicking on hyperlinks. In short, this is a typical web experience.

Note that our HTML formatters are optimized for small screens. (We used an HP 660LX for our experiments.) Thus, the same output will suffice for both PCs and PDAs. However, since we optimize for a

PDA, we expect that an alternative formatter could produce a richer experience on a full-size PC.

Our prototype speech client is a Java application, which means it is typically run from a command-line, rather than from within a browser window. Once the application is started, it works much like the HTML client.

For voice recognition, we use IBM's ViaVoice 98 Executive Edition(tm), supplemented by speech enablement for Java. That package maps the JSML API onto ViaVoice. First, the user must authenticate himself, which he does by speaking the command: login <username>.

The user must then complete the authentication by validating that he is the user he asserted he is. If the user has access to a keyboard, or even a numeric keypad, he can supply a password or PIN as he would in a text situation. Alternatively, smartcards or biometric identification equipment can be used, where available.

However, in a speech-only environment, these solutions are not practical. Instead, we use a challenge/response mechanism. The user preconfigures a number of questions and answers. When he logs in, the system selects one of these questions at random. His response is compared to the answer previously provided. While completely satisfactory, it will suffice for the purposes of our prototype.

One additional complication is that the quality of free-form voice recognition is still quite poor. Thus, were we to rely on a free-form answer to the password challenge, we'd often get cases where the user's correct response was recognized as an incorrect word.

To improve recognition quality, we use a restricted grammar. As described in the Java Speech API homepage, the speech recognition engine can be supplied with a BNF-like grammar that defines which phrases are acceptable. Our infrastructure populates the grammar with the correct answer and a number of incorrect

choices. When a word (or phrase) is spoken, the engine determines whether or not it matches a token in the grammar. If so, we pass that token to the server for the comparison with the correct answer; if not, we report failure to the server.

For example, if the user registered the question, "What is your dog's name?," and the correct answer is "spot," the system would also send down such incorrect responses as "tiger," "muffin," etc. A correct answer would be returned only if the recognition engine heard "spot;" failure would be returned if it heard "tiger" or "muffin," or if it could not match the response to one of the valid phrases.

In some cases, the user will not want to run the speech application on his local device. Yet, a user with only a screenless cell phone should still have access to his data sources. In such cases, we employ a dial-in proxy, as shown in Figure 3.

The user calls the proxy, which answers the call and runs our Java application. The connection between the user and the proxy is standard voice over cellular; the connection between the proxy and our infrastructure uses HTTP over the internet.

While we have not yet created a SpeechML client, it is worth discussing the differences between JSML and SpeechML. JSML is a class library that exports speech functions in a Java environment. Much like the structure of GUI programs, JSML programs typically have a main routine that waits for events from the (speech) UI.

In contrast, SpeechML programs are typically structured more like a series of web pages. Each "page" consists of three types of components: spoken "prompts," a list of valid responses to each

prompt, and actions that are to be performed for each valid response. Prompts often have a form such as:

Say 'one' to get your urgent messages

Say 'two' to get your non-urgent messages

Valid responses to these prompts are, of course, 'one' and 'two.' Actions tell the SpeechML browsers how to react to each spoken response. As in a standard web environment, actions are typically

hyperlinks. In our example above, the linked pages might contain the urgent and non-urgent messages.

Since SpeechML is quite similar to HTML, SpeechML output from our formatters would be quite similar to HTML output. Of course, additional attention must be paid to ensure that the spoken prompts are easily distinguished auditorially, and that valid responses can be distinguished by the speech recognition system.

We've discussed an infrastructure that allows multimodal access to a variety of data sources. Our standard architecture for adding data sources reduces the impact such additions have on the rest of the system. Similarly, adding new modalities has no impact on either existing source or other existing clients.

Notes: (1) Recall that the Visitor pattern is useful when "the classes defining the object structure rarely change, but you often want to define new operations over the structure." This is exactly the situation with the DOM tree. The objects in the tree are standardized, and thus will change infrequently, but the operations to be performed change with each new desired transformation. See also Design Patterns, Gamma, et al., p.331.

2/13 - (C) IBM CORP 1993

AN - NNRD408143

TI - A Process for Customized Information Delivery

PUB - Research Disclosure, April 1998, UK

VOL - 41

NR - 408

TXT - IBM's recent announcement of an Internet-enabled car introduces possibilities for new mechanisms for information delivery. Here we describe a process for customized delivery of information to a person's automobile. The goal of the process is to allow the user to get relevant information delivered in a time-efficient manner. The process, in short, is simple: have the user's home PC surf the web for him gathering material; translate the material into audio format; send the audio to the car and store it; and have the car replay the audio.

- In more detail, the process is comprised of the following steps:

- 1) information gathering and filtering
- 2) audio production
- 3) delivery of the (audio) information to the car
- 4) storage of delivered material for later replay
- 5) replay

The information gathering uses standard web techniques. A user specifies topics of interest (e.g., U.S. politics, soccer, middle east), and his computer stores these in a profile. Overnight, it uses standard search engines to locate pages matching the search criteria. It then downloads only pages from designated sites (e.g., CNN, NY Times,

ESPN)

created since the last search.

Alternatively, the user can use existing customized news services, such as MyYahoo (<http://my.yahoo.com>) to gather the news.

The web pages are then run through a speech synthesizer to create an audio file. Several speech synthesizers are sold commercially.

The information is then transmitted to a receiver in the designated vehicle. Transmission can use one of several techniques, including a cellular telephone call, or more economically, a 900MHz transmitter and receiver. (900MHz telephones, which contain a transmitter -- handset -- and receiver -- base -- can be purchased for under \$70.) Other transmission mechanisms are possible.

The information is then stored by the car either in RAM or a writeable media such as a writeable CD or a hard drive. IBM's Bamba audio format

(<http://www.alphaworks.ibm.com/examples/bambaforjava/example.html>) requires approximately 6 Kbits/sec to transmit audio, so (e.g.) 30 minute

of recording requires about 10Mbit or under 1.5MB. Each megabyte of commodity RAM is extremely inexpensive, so the storage is economically feasible.

That information can be replayed by the user upon request.

Note that if the car actually contains a processor -- as in the Internet car -- the information delivery and audio generation steps can

be swapped. In fact, the need for the user's PC can be eliminated if the

user is willing to allow the car to perform every step. Since the cost

of connecting a mobile device to the Internet is still rather high

via

cellular, and 900MHz is low bandwidth, such a tradeoff is not currently economical in most cases.

Similarly, after the information gathering step, the information can be transmitted to the car's computer as text, and the

audio can be synthesized by the car's computer itself.

Other variations on the process will be obvious to one skilled in the art.

3/13 - (C) IBM CORP 1993

AN - NN971223

TI - Method and Apparatus of Integrating a Personal Computer, Televisions, and Telephones into a Low-Cost Home Network

PUB - IBM Technical Disclosure Bulletin, December 1997, US

VOL - 40

NR - 12

PG - 23 - 24

TXT - This document contains drawings, formulas, and/or symbols that will not appear on line. Request hardcopy from ITIRC for complete article.

Disclosed is a low-cost method and apparatus to solve the problem of inconvenient access to the Personal Computer (PC). A home

network (36) (shown in the Figure) consists of a PC (2), a phone-line switch (4), a telephone (8), a wireless pointing device (26), a Television (TV) (16), a pair of TV signal transmitter (10) and receiver (12), a pair of audio signal transmitter (28) and receiver (30), a speaker (34), control signal transceivers (18) and (20), and a home appliance (22). The key features of this network are incoming message alert, remote access to the PC (2) with the telephone (8), and web surfing and Compact Disk-Read Only Memory (CD-ROM) game playing on a TV (10).

There are two methods to implement the incoming message alert. The first one uses the speaker (34); once the PC (2) receives a voice message or an e-mail, it drives the audio signal transmitter (28) to send a voice signal to the receiver (30) through wires or via radio frequencies. The speaker (34) can then announce the type of message and name of the person who should receive it. The second method uses different ring patterns to identify different messages. For instance, one ring represents an incoming e-mail for person A and two rings indicates an incoming e-mail for person B. The different ring patterns can be generated by activating the phone-line switch (4) to ring the phone or a sound generator, such as a wireless door chime.

For remote access to the PC (2), a user issues commands and listens to voice feedback from the PC (2) through the phone-line switch (4) and telephone (8). By default, the telephone (8) is connected to an external phone line until the off-hook signal and a specific keypad stroke pattern (for example, ##1) are detected. After this event occurs, the phone-line switch (4) connects the telephone (8) to the PC (2) which executes the tasks for voice commands and text-to-speech functions.

The phone-line switch (4) enables the user to retrieve voice and electronic mails, get stock quotes from web, and control the home appliance (22) in any place with a telephone (8).

For web surfing and CD-ROM game playing, visual feedback is essential. A pair of low-cost TV transmitter (10) and receiver (12) sends the image generated by the PC (2) to the television (16); the transmission media can be radio frequencies or residential wires. With a wireless pointing device (26) and the telephone, the user can comfortably surf the net or play interactive CD-ROM games in his living room while the PC (2) is in somewhere else, for example, the study room.

4/13 - (C) IBM CORP 1993

AN - NN9502433

TI - Notification of Availability of Office Equipment through Telephone Call

PUB - IBM Technical Disclosure Bulletin, February 1995, US

VOL - 38

NR - 2

PG - 433 - 434

TXT - This document contains drawings, formulas, and/or symbols that will not appear on line. Request hardcopy from ITIRC for complete article.

This article describes a mechanism that notifies a user of office equipment, with a telephone, that the equipment becomes available.

Office equipment, such as copiers and OHP foil makers, are usually shared among persons in groups, and located in places which may be far from desks of some users. This sharing normally increases the cost-effectiveness by increasing utilization of the equipment. However, in terms of the utilization of human resources, this sharing usually increases time wasted for doing useless activities. For instance, if a person brings sheets of paper to a copier and finds it busy (used by another person), he has to wait for the copier to become available, or return to his desk and come back again after some time. Or, he may have to ask the current user to let him know when the copier becomes available. (Here, a copier is used just as an example of office equipment, and other equipment can be applicable.)

With the mechanism in this article, a user need not wait at the copier until it becomes available, nor ask the current user to let him know when it becomes available. Instead, he enters his telephone number so that the copier informs him of its availability later by a phone call.

Fig. 1 shows an overall configuration in which telephones and copiers are connected through PBX (private branch exchange) network.

Although telephones and copiers are shown to make phone calls each other in Fig. 1, only copiers make a phone call to a telephone in this article. A copier notifies a person of availability with synthesized or recorded voice messages.

Fig. 2 shows internal components (devices) of typical office equipment with this mechanism, where Functional Component and User Interface are common to conventional office equipment, and the rest of components are added for this mechanism: (1) a device that places a telephone call, (2) a device that "speaks" messages, which are synthesized or recorded in advance, (3) a device that answers a telephone call, (4) a device that recognizes DTMF tones as requests or commands, (5) a device that process commands issued as DTMF tones or from User Interface, (5) a device that monitors the status of the equipment, makes a phone call to the current user, and drives the Speak device to notify the status change with voice messages. Components (3) and (4) are shown for completeness, and are not used in this mechanism.

5/13 - (C) IBM CORP 1993

AN - NA9002137

TI - Use of a Voice Communications Adapter As a Flexible Communications Method.

PUB - IBM Technical Disclosure Bulletin, February 1990, US

VOL - 32

NR - 9A

- TXT - - This article describes a technique by which a personal computer (PC)- based automation controller can communicate in a flexible manner with remote personnel or machines.
- Automation equipment is typically controlled by industrial computers. These PC-based systems provide an easy, flexible, strategic, as well as common, programming and hardware architectural environment. The voice communications adapter is a card for the PC that allows the PC to recognize or synthesize the human voice as well as controlling a telephone line and acting as a modem.
 - Before a tool is shipped from a plant, it is usually operated for a period of time to produce and eliminate any early life failures. In order to meet expected production schedules it is necessary to consider running the tools unattended. The problem, however, is the possibility of soft failures that could disrupt the process early. In response to this particular problem, the controller is provided with subroutines that enable the program to select personnel names and numbers from a file, and dial that number in an effort to locate and convey the current error status of the machine.
 - The voice communications adapter disclosed herein provides for the control and monitoring of a telephone extension. Among the functions that are provided is the ability to take the handset off hook, identify a dial tone, dial a number, check for a carrier from a modem, check for a possible response from a human voice, generate speech over the phone from text strings, and identify digit tones. By using these functions, a means is provided to the controller of a given tool to access a file containing necessary data to locate and identify a remote person or machine by telephone. Exception handling is provided by virtue of the ability of the software to obtain the status of the line and to monitor for given conditions. The application could be programmed to transmit ASCII data or speech in a flexible manner based on program logic and with feed back and or verification from a party using the tones generated by a TOUCH-TONE* phone. The intended result of the application described in this disclosure is to provide a set of subroutines that are enabled by the PC operating system to use the features of the voice communications adapter to communicate in a flexible manner with remote locations using an existing network (the telephone). Initial code was written in C to support the above functions.
 - The drawing shows in block diagram a flexible voice communication system for a manufacturing environment.

* Trademark of AT&T Co.

6/13 - (C) IBM CORP 1993

AN - NN8903220

TI - Spoken Nickname Recognition Telephone Dialer

PUB - IBM Technical Disclosure Bulletin, March 1989, US

VOL - 31

NR - 10

PG - 220 - 221

TXT - - Disclosed is a telephone system, based on, known technologies, which automatically dials an intended individual's number in response to a user's spoken request for that person by nickname. Referring to the block diagram in the Figure, the system operates as follows: A user wishing to call a person named "Joe Guy" employs handset 1 to verbally make the request "call Joe". This vocal message is received and digitized by Voice Input Processing module 2. Controller 3 then

passes the digitized request to Pattern Matching Processing module 4. This module searches the pre-store data base in Voice Match Data and System Messages module 5 for a nickname matching the current request for "Joe". If a match is found, the "Joe" data record is copied to the Controller. The Controller, using this data and Voice Synthesis Processing module 6, generates the spoken message to the user "calling Joe Guy". No verbal response from the user within some predetermined interval indicates confirmation. The Controller then transfers the numeric telephone number data from the "Joe" data record to Network Access Processing module 7 which proceeds to dial the number and connects the called line to handset 1. The telephone number may be for a local PBX extension, tieline, WATS line, outside local or long distance exchange line, or any other line that the user could dial himself. If the confirmation statement generated in the above case is "calling John Jones", a misunderstanding has occurred. The user repeats his request "call Joe", and the system tries again. If Pattern Matching Processing module 4 cannot find a match to the requested nickname, it may be an indication that the request has been misunderstood. The Controller then generates the spoken message to the user "repeat request". A second successive failure may indicate that "Joe" is not on record. Given the above functional capability, it is apparent that the system is not limited to the straightforward usage described above. The data record for "Joe" can include multiple telephone numbers to be selected from depending on the date or time of day, (office, factory, home, other known locations), (network cost differences vs. time, network availability, etc.). The system can be used as a directory, permitting a user to call from a remote location, verbally request "number for Joe" and receive a system-generated voice response, "number for Joe Guy is five five five seven three one two". Considering the possibility of distinguishing different voices, "Joe" may be different people to different users.

7/13 - (C) IBM CORP 1993

AN - NN8707656

TI - Enhanced Personal REMINDER Facility

PUB - IBM Technical Disclosure Bulletin, July 1987, US

VOL - 30

NR - 2

PG - 656

TXT - - This system provides an effective mechanism for generating and delivering a computer-controlled personal reminder regardless of what application program is executing. The system is based on an independent time-keeping process. On the IBM PC, under PC-DOS, this process can be a device driver as described below. Under other operating systems on the IBM System/370, the process can be performed by a separate virtual machine or a task running without an attached terminal. The independent time-keeping process (REMINDER DEVICE DRIVER) is used to control the storing and delivery of reminders. The device driver is a piece of virtual hardware that acts like an input/output device. An application program, such as a calendar, can write reminders to the REMINDER DEVICE DRIVER which then acts as a storage device and clock watcher. At the appointed time, the REMINDER DEVICE DRIVER verifies the best means to deliver the reminder, e.g., signal to the calendar program, audible alarm,

computer synthesized voice, telephone call, or other appropriate means. Each time a new reminder mechanism is added to the system, a message must be written to the REMINDER DEVICE DRIVER so that this means can supersede the current means as the best available. In addition, each time a mechanism is removed from the system, a corresponding message restores the prior best means for delivery. The novelty here is in connecting the device driver to an application that maintains its primary data at another network node or a host computer. In addition, the connection of the alarm capability to a more natural user interface (such as a telephone or computer synthesized voice) enhances the REMINDER DEVICE DRIVER to make it more usable and understandable.

8/13 - (C) IBM CORP 1993

AN - NN8507897

TI - Universal Tone Transmitter/Receiver Module

PUB - IBM Technical Disclosure Bulletin, July 1985, US

VOL - 28

NR - 2

PG - 897 - 899

TXT - - This article describes a universal tone receiver/transmitter module which provides a telephone switching system with one module for all tone signaling. The application of all digital technology offers many advantages over present methods of telephony-oriented tone generation and detection. Adaptation to a telephone company (telco) tone plan anywhere in the world requires no physical or electrical adjustments, as the CPU can be programmed to update the tone plan parameters. Each tone plan may also have many subsets of tones. For example, in the USA, tone subsets include the dual-tone multi-frequency (DTMF) plan, the multi-frequency (MF) plan, and the progress tone plan. Other countries have their own unique tone plans. The tone module of this disclosure easily adapts to these varied tone plans with only software changes. Other private automatic branch exchanges (PABXs) use multiple module types to perform these tone signalling functions. This older technique of multiple modules requires many different types of spare modules to maintain a PABX installation. It also requires that a different set of tone modules be used whenever the installation site uses a non-standard tone plan.

The present universal tone transmitter/receiver module reduces these varied tone module requirements to a single module type. Another advantage is in the predominant use of digital technology which facilitates automated manufacturing and testing. The universal tone transmitter/receiver module disclosed herein is designed to be a function module in a larger telecommunication system such as a PABX as shown in Fig. 1. This universal tone transmitter/receiver module can be used in any application where telephony signalling information is processed in the frequency range of 200 to 4000 Hertz. The function of this module is to synthesize and analyze audio waveforms.

All data for the generation of audio signals is down-loaded from the central processing unit (CPU) at system initialization and can be updated as the needs of the system change. Types of data involved in this audio generation include: frequency or frequencies, amplitudes, phase angles and duty cycles. Detection of audio energy involves determining the spectral components present, their related amplitudes

and phases. The types of functions this module can perform include: DTMF signalling, MF signalling, progress tone signalling, modem operations, and voice recognition/synthesis. The present tone module has eight input ports. These input ports are connected to a switching module which connects them to various telecommunication ports as the system requirements dictate (Fig. 1). The incoming audio waveforms are multiplexed and then converted to digital sample sets using an analog-to-digital converter. The digital sample sets are then stored in the memory section of the CPU. The signal processor then performs the required operations as defined by the CPU on these sample sets in order to analyze the characteristics of the audio waveforms. The results of the computations are then stored in the memory section awaiting transfer to the CPU. The universal tone transmitter/receiver module subsystem is shown in block diagram in Fig. 2 and includes a microprocessor-controlled bus interface 1 to the next level processor, a digital signal processor 2 to analyze and synthesize audio waveforms, an analog/digital converter 3 to translate between the digital and analog domains, a multiplexer 4 to time division multiplex the signal processor among multiple ports, a programmable hardware section 5 to generate commonly used tones under the control of the microprocessor, and a memory section 6. The tone module has six output ports that are under direct digital control of the bus interface microprocessor 1. The CPU downloads digital representations of the audio waveforms to the tone module. The digital instructions are decoded, and the one waveform cycle is repeatedly sent to the digital/analog converter 3. Optional parameters for the transmission of the waveform include time durations, on/off cycle times and alternating between two different waveforms. The signal processor 2 has direct control of two output ports. These ports are for infrequently used messages and d messages and speechsynthesis. To output on these ports, the CPU transmits to the tone module a digital data string containing the waveform information. The signal processor composes the desired output string in a digital format and sends it to the digital/analog converter. The analog waveform is demultiplexed to one of the two output ports. The synthesized audio waveforms at the output ports of this tone module are resources for the PABX system to utilize as the network requires.

9/13 - (C) IBM CORP 1993

AN - NN85014989

TI - Optimal Retention Delay in the Receiver of a Digital Voice Network

PUB - IBM Technical Disclosure Bulletin, January 1985, US

VOL - 27

NR - 8

PG - 4989 - 4990

TXT - - At the receiving end of a digital voice packet transmission network, voice synthesized signals have to be re-synchronized due to packet transmission delay jitter. To avoid these distortions, instead of being fed into the synthesizer as they arrive, the packets are fed into a buffer register. They are then fetched out of the buffer sequentially after a fixed length retention delay greater than the maximum transmission jitter expected. This retention delay is reset after each long speech pause (e.g., greater than a predetermined length value). The transmission delay jitter is thus absorbed by the

long speech pauses, while effective speech signal synthesis is not affected. This method may however penalize the system by unduly affecting slow packets. A solution is proposed here to avoid these drawbacks. After a long pause, the first incoming packet is systematically fed into a queue buffer at a position from which it would be extracted for further being used for voice synthesis after an initial retention delay greater than the maximum expectable network delay for the considered packet channel used. The packet is then systematically shifted toward the buffer output at a rate equal to the packet transmission rate. The subsequent packets are then normally fed into the buffer register. The queue buffer position corresponding to the initial retention delay is made to represent a buffer threshold. In operation, the threshold position is permanently monitored. Assuming the incoming packets are all introduced above the threshold, the threshold is made to shift one position lower at the first incoming packet following the next long pause. This operation may be repeated until a first packet is made to occupy a queue position below the threshold, or vice-versa. The above method could be implemented using different means. For instance, a counter CPTR could be incremented upon each packet being received. Then a device could be used for computing an increment parameter (DELTA) for each incoming packet, according to: $DELTA(n) = CPTE(n) - CPTR(n)/\text{mod } 8$ wherein - n represents the nth received packet, and - CPTE(n) is a 3-bit number assigned sequentially to each transmitted packet and coded within the packet frame. Then the queue packet (PP) position where the incoming nth packet should be fed into would be derived from the previous packet position through: $PP(n) = PP(n-1) - DELTA(n-1) + DELTA(n)/\text{mod } 8$ The first packet position would be $PP1 = 3$.

10/13 - (C) IBM CORP 1993

AN - NB84124356

TI - Carbon Microphone Inverse Filtering

PUB - IBM Technical Disclosure Bulletin, December 1984, US

VOL - 27

NR - 7B

PG - 4356 - 4357

TXT - - In a digital voice network using low bit rate voice compression

techniques, a critical problem may appear due to the distortions introduced by the combination of carbon microphones and telephone lines. More specifically, when the speech coder is located close to the speaker, one can use a dynamic microphone and a telephone line with quasi-flat frequency response, thus ensuring that a clean digital speech signal is available at the input of the speech coder, which results in a good quality synthesized speech. This solution is

currently applied in small configurations. However, the price to pay in case of a large digital voice network would be too high since in

this case each customer would have its own voice digitizer coupled with a special telephone set. For large networks, a solution consists in sharing a voice coder for several users which can get connected through the public switch telephone network, using already installed

carbon microphones. In this case, the speech coder is located close to a PBX. While presenting the best trade-off in terms of

implementation cost, this solution generally results in a poorer speech quality at the output of the synthesizer. This is due to the distortions added to the speech signal by the telephone line and by the carbon microphone. Low bit rate speech coders are generally very sensitive to these distortions. One can however reduce these distortions by preprocessing the input speech signal before bit rate reduction. In this preprocessing, the line distortions are assumed to be linear, that is, the line is supposed to mainly introduce a frequency attenuation on the signal (band-pass filter). The carbon microphone is modeled as a combination of linear distortion and non-linear distortion. It is assumed that the non-linear distortion consists in the corruption of the speech signal with a noise proportional to the speech envelope. Thus, it is possible to proceed to an adaptive filtering of the input speech signal. This filtering makes use of a comb filter adapted to the pitch period of the input speech, where the coefficients are adapted from the energy of the input speech signal. The global linear distortion due to the telephone line and to the carbon microphone is compensated by prefiltering the input speech with an adaptive inverse filter (basically a band-stop filter), the coefficients of which are adapted from the frequency analysis of the input speech signal.

11/13 - (C) IBM CORP 1993

AN - NA84034947

TI - Pseudo Hangover Synthesis

PUB - IBM Technical Disclosure Bulletin, March 1984, US

VOL - 26

NR - 10A

PG - 4947

TXT - - In a digital voice network wherein N conversations are to be transmitted over C equivalent channels ($N > C$), the channel assignments are based on voice activity detection. The channels are assigned to active sources only. In other words, silences are not transmitted. However, it is difficult to detect the end of a talk-spurt, because the voice activity does not stop instantaneously. To provide smooth restitution, the voice sources are still considered active after each talk-spurt during a "hangover" time. Such hangovers do, however, increase the load of the network. A solution is proposed here to minimize the hangover load by not transmitting any voice signal during the hangovers and substituting for it a pseudo hangover voice signal generated at the receiving end of the network. The proposed solution deals with stopping voice transmission as soon as the voice source energy drops under the voice activity detection (VAD) level, and to synthesize the pseudo hangover. The synthesized voice is provided by an attenuated reconstruction of the received voice signal prior to VAD indicating the occurrence of a silence.

12/13 - (C) IBM CORP 1993

AN - NN83014474

TI - Voice and Data Transmission. January 1983.

PUB - IBM Technical Disclosure Bulletin, January 1983, US

VOL - 25

NR - 8

PG - 4474 - 4475

TXT - 2p. This is a voice and data transmission system. The

proposed architecture is based on the fact that two main functions have to be performed, i.e., voice compression, and voice and data multiplexing.

- Voice compression consists in reducing the voices PCM data rate from 64 Kbps to 7,200 bps using split band and dynamic allocation of quantizing bit techniques. More particularly, the 64 Kbps is fed into a Voice Excited Predictive Coder (VEPC) which derives PARCOR parameters (K) therefrom; these parameters are used to derive a redundancy-free residual signal from the original vocal signal. The residual signal is fed into a low-pass filter which derives therefrom a band-limited or residual baseband signal together with information relating to the energy (E) contained within the removed high frequency bandwidth. The residual baseband is in turn split into p sub-bands, the contents of which are requantized using dynamic bit rate allocation techniques. It should be noted that the above speech analysis operations are performed over blocks of samples 20 ms long. The residual baseband is thus processed in block companded PCM (BC PCM). Each block of samples provides one or two E, a block of eight PARCOR coefficients, and requantized samples such that the overall bit rate is limited to 7,200 bps.
- Conversely, on the receiving side of the transmission network, energies, PARCORS and samples will have to be recombined for synthesizing the original speech signal.
- Both analysis and synthesis of the speech signal are performed using a single tributary microprocessor MP1. Every 125 microseconds, the PCM coded data is serially loaded into a shift register SR1, and then transferred into an input voice buffer VBI. The status of VBI Z. contents is indicated to MP1 by setting one bit in the status buffer STAT. Interrupt must be performed within 125 microseconds following the status change.
- Incoming voice data information is buffered in MP1 for 20 ms, and then compressed to 7,200 bps using the above-mentioned algorithm.
- Also, every 125 microseconds, output buffer VB2 is loaded by MP1 and then transferred into output shift register SR2.
- The second main function of the network, i.e., voice and data multiplexing, is then performed. The voice data transmission network is mastered by a microprocessor MPo. This microprocessor controls the multiplexing of the 7,200 bps speech-originating data with a 2,400 bps data channel, over a 9,600 bps data link using a 9,600 bps modem.
- More particularly, at MPo request (one bit set in the status buffer STAT and its associated interrupt), the compressed data is transferred into a MP1 output buffer in burst form on a 16-bit word basis. Similarly at MPo request, a MP1 input buffer, when full, must be read by MP1 for decompression, and 64 Kbps PCM sample restitution. This operation occurs on a 20 ms basis.
- The compressed voice data transfer from the MP1 output buffer to the 9,600 communication link and from this link to MP1 input buffer is performed on a byte by byte basis through a full duplex communication adapter CCA1.
- A similar communication adapter (CCA2) is used to interface the 2,400 bps data channel.

13/13 - (C) IBM CORP 1993

AN - NN71043356

TI - Spectrum Flattening in Vocoders. April 1971.

PUB - IBM Technical Disclosure Bulletin, April 1971, US

VOL - 13

NR - 11

PG - 3356

TXT - 1p. In a conventional base-band vocoder synthesizer, the speech quality is improved by the so called 'spectrum flattening' feature. This is performed by using on each of the excitation channels a premodulation network made of a band-pass filter BPF_i followed by a clipper CL_i , $i=1, 2, \dots, n$ with generally $n \approx 15$.

The clipping operation generates odd harmonics which could fall within another channel bandwidth, and, therefore, could distort the output signal provided by the summing in stage Sigma. These harmonics must be removed. In conventional synthesizers the removal

is obtained through use of a second band-pass filter on each channel (post-modulation network).

- In order to facilitate a digital implementation of the synthesizer, it is of high interest to remove the post-modulation filters. A solution is provided through use of a frequency shifting operation. A carrier frequency F_0 is used to modulate the excitation signal in stage M1 before driving the premodulation filters. The frequency bandwidth of each BPF_i is chosen such that the lower sideband generated by modulator M1 is removed. In other words the excitation frequency spectrum is shifted towards the upper frequency.

Therefore, the odd harmonics generated by the clippers are also shifted the same way. The frequency F_0 is such that these shifted harmonics fall outside the bandwidth of the vocoder channels and, therefore, cannot interfere with a useful signal provided by any other channel. Thus, the post-modulation filters, not shown, are needless and can be replaced by a single low-pass filter LFP_1 , it is easy to implement and remove the odd harmonics. The synthesized signal is demodulated by M2 and low-pass filtered to shift the speech signal back to the audio band.

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File 349:PCT Fulltext 1983-2000/UB=20000921, UT=20000908

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Set	Items	Description
S1	1005	((TEXT? ?(2W) SPEECH))(5N) SYSTEM? ? OR TTS
S2	971	TEXT? ? (2N)(TRANSFORM? OR CONVERT? OR CONVERSION? OR SYNT- HES? OR (CHANGE? OR TURN?)(2N)INTO)(5N) (SOUND OR AUDIO? OR V- OICE? OR SPEECH)
S3	1785	S1 OR S2
S4	22	S3 (10N) ((WEB OR NETWORK OR W3 OR INTERNET OR INTRANET)(- 5N)(SERVER? OR SITE?) OR WEB() PAGE?)
S5	217	AUDIO(2W)(WAVEFORM? OR WAVE()FORM?)
S6	4522	(PROSOD? OR ACCENTUAT? OR INTONATION?)
S7	4873	(SPEECH OR VOICE) (2N) (SYNTHE? OR GENERAT?)
S8	5710	CONCATENAT?
S9	2437	(SPEECH? OR SOUND? OR VOICE)(2N)(FRAGMENT? OR SAMPL?)
S13	122635	(PITCH? OR DURATION OR APTITUDE OR (ATTACK OR DECAY)(2N) E- NVELOP? OR (SYNTHE? () INSTRUCT?))
S15	805	((TEXT? ?(2W) SPEECH))
S16	36	S15 (5N) SERVER?
S17	61	S15 (S)S6
S18	2	S17(S)S9
S19	34	S17(S)S13
S20	7	S17 AND SERVER?
S22	3	S16 (S) (S5 or S6 OR S8:S13)
S26	437	(SYNTHE? (3N) INSTRUCT?)
S27	2	S26 (S)S15
S28	10292	SYNTHE? (3N) (INSTRUCT? OR COMMAND? OR DIRECT?)
S29	7	S28(S)S15 NOT S27

full fr.
patents

RA

20/3,AB/1 (Item 1 from file: 348)

DIALOG(R)File 348:European Patents

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00602713

Text-to-speech processor, and parser for use in such a processor

Prozessor zur Umwandlung von Daten in Sprache und Ablaufsteuerung hierzu

Processeur de conversion texte-parole et utilisation d'un analyseur dans un tel processeur

PATENT ASSIGNEE:

Canon Information Systems, Inc., (1553870), 3188 Pullman Street, Costa Mesa, CA 92626, (US), (Proprietor designated states: all)

INVENTOR:

Luther, Willis J., 5 Spicewood Way, Irvine, California 92715, (US)

LEGAL REPRESENTATIVE:

Beresford, Keith Denis Lewis et al (28273), BERESFORD & Co. High Holborn 2-5 Warwick Court, London WC1R 5DJ, (GB)

PATENT (CC, No, Kind, Date): EP 598598 A1 940525 (Basic)

EP 598598 B1 000202

APPLICATION (CC, No, Date): EP 93309147 931116;

PRIORITY (CC, No, Date): US 978487 921118

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G06F-003/16

ABSTRACT EP 598598 A1

A text parser (34) for a text-to-speech processor accepts a text stream and parses the text stream to (36) detect non-spoken characters. A text generator generates pre-designated text sequences in response to non-spoken characters, such as special character sequences or character sequences which match format templates. A speech command generator (37) generates speech commands in response to detecting of non-spoken characters such as non-spoken characters which affect text style, font, underlining, etc. A text-to-speech converter (26) converts spoken text parsed by the parser and text generated by the text generator into speech, the text-to-speech converter being operable in response to speech commands generated by the speech command generator. According to the invention, it is not necessary to pre-process text files in preparation for text-to-speech conversion, and arbitrary files which contain both spoken and non-spoken characters may be converted easily. (see image in original document)

ABSTRACT WORD COUNT: 146

NOTE:

Figure number on first page: 2

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	200005	2225
CLAIMS B	(German)	200005	1935
CLAIMS B	(French)	200005	2666
SPEC B	(English)	200005	4719
Total word count - document A			0
Total word count - document B			11545
Total word count - documents A + B			11545

20/3,AB/2 (Item 2 from file: 348)

DIALOG(R)File 348:European Patents

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00424387

Text-to-speech system having a lexicon residing on the host processor.
Text-zu-Sprache Übersetzungssystem mit einem im Hostprozessor vorhandenen
Lexikon.

Système de conversion texte/parole comportant un lexique résident dans le
processeur principal.

PATENT ASSIGNEE:

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PATENT (CC, No, Kind, Date): EP 429057 A1 910529 (Basic)

APPLICATION (CC, No, Date): EP 90122169 901120;

PRIORITY (CC, No, Date): US 439240 891120

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G10L-005/04; G06F-003/16;

ABSTRACT EP 429057 A1

A text-to-speech system is provided having a host system operable to perform a text-to-speech application program. The host system includes a memory storing the lexicon for a separate text-to-speech device. By using the host system to contain the lexicon, a sufficient amount of memory is made available. Therefore, a very complex lexicon can be provided and more information made available to the voice synthesizer on the text-to-speech device. This partitioning of the text-to-speech system between the host system and the text-to-speech device allows the computation-intensive processes to be performed on the text-to-speech device while providing a large memory to contain the lexicon information.

ABSTRACT WORD COUNT: 107

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPABF1	430
SPEC A	(English)	EPABF1	2087
Total word count - document A			2517
Total word count - document B			0
Total word count - documents A + B			2517

20/3,AB/3 (Item 1 from file: 349)

DIALOG(R)File 349:PCT Fulltext

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00709639

AUTOMATIC INQUIRY METHOD AND SYSTEM

PROCEDE ET SYSTEME D'INTERROGATION AUTOMATIQUE

Patent Applicant/Assignee:

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Inventor(s):

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Patent and Priority Information (Country, Number, Date):

Patent: WO 0022549 A1 20000420 (WO 200022549)

Application: WO 99EP7032 19990922 (PCT/WO EP9907032)
Priority Application: EP 98203423 19981009
Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE
Publication Language: English
Filing Language: English
Fulltext Word Count: 7592

English Abstract

A system (10) for automatically responding to an inquiry from a user comprises a dialogue manager (50), which executes a machine- controlled human-machine dialogue to determine a plurality of pre- determined query items. The dialogue manager (50) retrieves a plurality of information items from a storage (52) in dependence on the query items. The system further comprises a presentation manager (90) which determines an intention of the user, reflecting a preferred way of presenting the information items. The presentation manager (90) selects a presentation scenario from a predetermined set of presentation scenarios (96) in dependence on the determined intention. At least one natural language phrase is generated to present the obtained information items according to the selected presentation scenario. A speech generator (60) verbally presents the generated phrase(s) to the user.

French Abstract

L'invention concerne un systeme (10) concu pour repondre automatiquement a l'interrogation emanant d'un utilisateur. Ledit systeme (10) comprend un gestionnaire de dialogue (50) qui execute un dialogue homme-machine commande par une machine, pour determiner plusieurs articles d'interrogation predetermines. Le gestionnaire de dialogue (50) extrait plusieurs articles d'information d'une memoire (52), en fonction des articles d'interrogation. Ledit systeme comporte egalement un gestionnaire de presentation (90) qui determine une intention de l'utilisateur, refletant une maniere preferee de presentation des articles d'information. Le gestionnaire de presentation (90) selectionne un scenario de presentation dans un ensemble predetermine de scenarios de presentation (96), en fonction de l'intention determinee. Au moins une phrase en langage naturel est generee de sorte que les informations obtenues soient presentees en fonction du scenario de presentation selectionne. Un generateur de parole (60) presente verbalement a l'utilisateur la ou les phrases generees.

20/3,AB/4 (Item 2 from file: 349)
DIALOG(R)File 349:PCT Fulltext
(c) 2000 WIPO/MicroPat. All rts. reserv.

00692528

VOICE BROWSER FOR INTERACTIVE SERVICES AND METHODS THEREOF
NAVIGATEUR VOCAL POUR SERVICES INTERACTIFS ET PROCEDES ASSOCIES

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Inventor(s):

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JOHNSON Gregory, JOHNSON, Gregory, 565 Iroquois Trail, Carol Stream, IL
60188, US

Patent and Priority Information (Country, Number, Date):

Patent: WO 0005708 A1 20000203 (WO 200005708)

Application: WO 99US16776 19990723 (PCT/WO US9916776)

Priority Application: US 9894131 19980724; US 9894032 19981002

Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES
FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU

LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA
UG UZ VN YU ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU TJ TM
AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM
GA GN GW ML MR NE SN TD TG

Publication Language: English

Filing Language: English

Fulltext Word Count: 21304

English Abstract

A voice browser to process a markup language document. A voice browser includes a network fetcher unit to retrieve information from a destination of an information source. A parser unit is communicatively coupled to the network fetcher to parse the retrieved information based on predetermined syntax. The parser unit generates a tree structure representing the hierarchy of the retrieved information. An interpreter unit and a state machine are also used. The method includes the steps of retrieving and parsing a markup language document to determine at least one user input, determining whether the user input corresponds to a predetermined grammar, and using the predetermined grammar when the user input corresponds to the predetermined grammar. The method of determining a grammar is based upon phonetic rules and pronunciation. The grammar is sent to a speech recognition engine and compared to a user input.

French Abstract

La presente invention concerne un navigateur vocal capable de traiter un document HTML. Un tel navigateur comporte un module de recherche reseau permettant de retrouver une information en provenance d'une destination d'une source d'information. Un module d'analyse est couple communiquant au module de recherche reseau de facon a analyser l'information retrouvee en fonction d'une syntaxe definie. Le module d'analyse genere une arborescence representant la hierarchie de l'information retrouvee. Un module interprete est couple communiquant au module de recherche reseau de facon a traiter le document HTML. Un automate fini est couple communiquant au module interprete et au module d'analyse. Un procede selon l'invention consiste a retrouver un document HTML, a analyser le document HTML a la recherche d'au moins une entree utilisateur, a determiner si cette entree utilisateur correspond a une grammaire definie, et a utiliser cette grammaire definie si l'entree utilisateur consideree correspond a la grammaire definie. Le procede consiste enfin a reconnaitre la grammaire d'apres des regles phonetiques definies et la prononciation lorsque l'entree utilisateur consideree ne se trouve pas dans la grammaire definie, a envoyer la grammaire a un moteur de reconnaissance vocale et a comparer a une entree utilisateur la grammaire.

20/3,AB/5 (Item 3 from file: 349)

DIALOG(R)File 349:PCT Fulltext

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00692477

MARKUP LANGUAGE FOR INTERACTIVE SERVICES AND METHODS THEREOF

LANGAGE DE BALISAGE POUR SERVICES INTERACTIFS ET PROCEDES ASSOCIES

Patent Applicant/Assignee:

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Inventor(s):

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JOHNSON Gregory, JOHNSON, Gregory, 565 Iroquois Trail, Carol Stream, IL
60188, US

Patent and Priority Information (Country, Number, Date):

Patent: WO 0005643 A1 20000203 (WO 200005643)
Application: WO 99US16777 19990723 (PCT/WO US9916777)
Priority Application: US 9894131 19980724; US 9894032 19981002
Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES
FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU
LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA
UG UZ VN YU ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU TJ TM
AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM
GA GN GW ML MR NE SN TD TG
Publication Language: English
Filing Language: English
Fulltext Word Count: 20939

English Abstract

The present invention (Fig. 6) relates to a voice browser to provide interactive services. A markup language document in accordance with the present invention includes a dialog element (2) including a plurality of markup language elements (3-26). Each of the plurality of markup language elements is identifiable by at least one markup tag. A step element (11) is contained within the dialog element. The step element includes a prompt element (4) and an input element (9). The prompt element (4) includes an announcement to be read to the user. The input element includes at least one input that corresponds to a user input. A method in accordance with the present invention includes the steps of creating a markup language document having a plurality of elements (3-26), selecting a prompt element (2), and defining a voice communication (14) in the prompt element to be read to the user. The method further includes the steps of selecting an input element (2) and defining an input variable to store data inputted by the user.

French Abstract

La presente invention (Fig. 6) concerne un navigateur vocal capable de fournir des services interactifs. Selon la presente invention, on dispose d'un document en langage de balisage qui inclut un element de dialogue (2) integrant une pluralite d'elements de langage de balisage (3-26). Chacun de ces elements de langage de balisage est identifie par une etiquette de balisage. L'element de dialogue renferme un element d'etape (11). Cet element d'etape comprend un element d'invite (4) et un element d'entree (9). L'element d'invite (4) comporte une annonce a faire lire a l'utilisateur. L'element d'entree comporte au moins une entree qui correspond a une entree utilisateur. L'invention concerne egalement un procede consistant a creer un document en langage de balisage comportant une pluralite d'elements (3-26), a selectionner un element d'invite (2), et a definir dans l'element d'invite a faire lire a l'utilisateur une communication vocale (14). Le procede consiste enfin a selectionner un element d'entree (2) et a definir une entree variable permettant de ranger les donnees introduites par l'utilisateur.

20/3,AB/6 (Item 4 from file: 349)
DIALOG(R)File 349:PCT Fulltext
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00692472

METHODS AND SYSTEMS FOR ACCESSING INFORMATION FROM AN INFORMATION SOURCE
SYSTEMES D'ACCES A L'INFORMATION PAR UNE SOURCE D'INFORMATION ET PROCEDES
ASSOCIES

Patent Applicant/Assignee:

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Inventor(s):

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Patent and Priority Information (Country, Number, Date):

Patent: WO 0005638 A2 20000203 (WO 200005638)

Application: WO 99US16780 19990723 (PCT/WO US9916780)

Priority Application: US 9894131 19980724; US 9894032 19981002

Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES
FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU
LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA
UG UZ VN YU ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU TJ TM
AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI CM
GA GN GW ML MR NE SN TD TG

Publication Language: English

Filing Language: English

Fulltext Word Count: 21375

English Abstract

The present invention relates to systems and methods to provide a user with information from an information source. A system in accordance with the present invention includes a communication node including a switch having at least one incoming line. An audio processing unit is communicatively coupled to the switch to receive incoming audio communications from the user and to provide outgoing audio communications to the user. A voice browser is communicatively coupled to the audio processing unit. The voice browser retrieves information from the information source and provides an output to the audio processing unit. The audio processing unit provides an outgoing audio communication to the user in response to the output. A method in accordance with the present invention includes the steps of receiving an audio input from a user associated with a destination of an electronic network, connecting to the destination based upon the audio input, and retrieving information associated with the destination. The method further includes the steps of processing the information to generate a voice communication, and providing the voice communication to the user.

French Abstract

La presente invention concerne des systemes et procedes permettant de fournir a un utilisateur de l'information a partir d'une source d'information. Un tel systeme comporte un noeud de communications comportant un commutateur pourvu d'au moins une ligne en entree. Un module de traitement audio est couple communiquant au commutateur de facon a recevoir les communications audio entrantes en provenance de l'utilisateur, et de facon a fournir a l'utilisateur des communications en sortie. Un navigateur vocal est couple communiquant au module de traitement audio. Ce navigateur vocal va retrouver l'information dans la source d'information, puis il realise une sortie a destination du module de traitement audio. En reaction a la sortie, ce module de traitement audio fournit a l'utilisateur une communication audio en sortie. Le procede de l'invention consiste a recevoir une entree audio en provenance d'un utilisateur associe a une destination d'un reseau electronique, a se connecter sur la destination en fonction de l'entree audio, et a retrouver l'information associee a la destination. Le procede consiste enfin a traiter l'information de facon a generer une communication vocale, puis a fournir a l'utilisateur cette communication vocale.

20/3,AB/7 (Item 5 from file: 349)

DIALOG(R) File 349:PCT Fulltext

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00658883

COMPUTER-BASED PATIENT RECORD AND MESSAGE DELIVERY SYSTEM

SYSTEME DE DOSSIERS INFORMATISES DE PATIENTS ET DE REMISE DE MESSAGES

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Patent and Priority Information (Country, Number, Date):

Patent: WO 9942932 A2 19990826

Application: WO 99IB192 19990204 (PCT/WO IB9900192)

Priority Application: US 9827125 19980220

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Publication Language: English

Filing Language: English

Fulltext Word Count: 3070

English Abstract

A Computer-based Patient Record (CPR) system includes user equipment devices which are configured for speech synthesis in response to speech markup language text and which are connected via a network to a middle tier of a **server** system. The CPR system further includes a message delivery facility for delivery of textual messages to any of pager, electronic mail, or voice mail (after **text to-speech** synthesis) message delivery vehicles. The **server** system accesses a user specific data store containing speech synthesis profiles which include **prosodic** information of the voices and speech of users, and message delivery profiles which specify which of the aforementioned message delivery vehicles are to be used and in what order. The stored speech synthesis information associated with an originator of a message and the stored message delivery information associated with the recipient of message are provided by the **server** to user equipment or a reminder generator to produce speech markup files containing information needed to synthesize the vocal and speech characteristics of the originator accompanied by delivery instructions reflecting the message delivery preferences of the recipient.

18/3,K/1 (Item 1 from file: 349)
DIALOG(R)File 349:PCT Fulltext
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00723688

SPEECH OPERATED AUTOMATIC INQUIRY SYSTEM
SYSTEME DE RENSEIGNEMENTS AUTOMATIQUE A COMMANDE VOCALE

Patent Applicant/Assignee:

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PHILIPS CORPORATE INTELLECTUAL PROPERTY GMBH, PHILIPS CORPORATE
INTELLECTUAL PROPERTY GMBH , Habsburgerallee 11, D-52066 Aachen , DE

Inventor(s):

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Patent and Priority Information (Country, Number, Date):

Patent: WO 0036591 A1 20000622 (WO 200036591)

Application: WO 99EP9263 19991129 (PCT/WO EP9909263)

Priority Application: EP 98204286 19981217

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Publication Language: English

Filing Language: English

Fulltext Word Count: 7191

Fulltext Availability:

Detailed Description

Detailed Description

... the question/confirmation statements from the dialogue

71 1=

mana(yer 50 in various forms, such as a (potentially prosodically enriched) textual form or as **speech fragments** . The **speech** generation subsystem 60 may be based on speech synthesis techniques capable of converting **text** -to-**speech** . The speech generation subsystem 60 may itself **prosodically** enrich the **speech fragments** or text in order to generate more naturally sounding speech. The enriched material is then transformed to speech output. Speech generation has been disclosed in...

...a sentence with certain system-specific voice characteristics (e.g. the voice of an actor) and one isolated utterance (his own) in between.

Preferably, the **prosody** of the input utterance is changed to correspond to the **prosody** of the entire statement. Via the interface 70 the speech output is provided to the user at the speech output interconnection 80. Typically, a loudspeaker...

18/3,K/2 (Item 2 from file: 349)
DIALOG(R)File 349:PCT Fulltext
(c) 2000 WIPO/MicroPat. All rts. reserv.

00709639

AUTOMATIC INQUIRY METHOD AND SYSTEM
PROCEDE ET SYSTEME D'INTERROGATION AUTOMATIQUE

Patent Applicant/Assignee:

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Inventor(s):

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Patent and Priority Information (Country, Number, Date):

Patent: WO 0022549 A1 20000420 (WO 200022549)

Application: WO 99EP7032 19990922 (PCT/WO EP9907032)

Priority Application: EP 98203423 19981009

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Publication Language: English

Filing Language: English

Fulltext Word Count: 7592

Fulltext Availability:

Detailed Description

Detailed Description

... 60 may receive the question/confirmation statements from the dialogue manager 50 in various forms, such as a (potentially prosodically enriched) textual form or as **speech fragments**. The **speech** generation subsystem 60 may be based on speech synthesis techniques capable of converting **text** -to-**speech**. The speech generation subsystem 60 may itself **prosodically** enrich the **speech fragments** or text in order to generate more naturally sounding speech. The enriched material is then transformed to speech output. Speech

22/3,IC,K/1 (Item 1 from file: 349)
DIALOG(R)File 349:PCT Fulltext
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00682227

**SYSTEM AND METHOD FOR DELIVERING ELECTRONIC MESSAGING TO MOBILE PHONES
SYSTEME ET PROCEDE POUR ACHEMINER DES MESSAGES ELECTRONIQUES A DES
TELEPHONES PORTABLES**

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Inventor(s):

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Patent and Priority Information (Country, Number, Date):

Patent: WO 9965256 A2 19991216

Application: WO 99US13183 19990610 (PCT/WO US9913183)

Priority Application: US 9888781 19980610

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04Q-007/00;

Publication Language: English

Filing Language: English

Fulltext Word Count: 10960

Fulltext Availability:

Detailed Description

Detailed Description

... the e-mail notification server; incoming client request logging by 0
Z11D z1-"" C.

the Voice Mail Notification Server; incoming client request lo(... ing
by **text -to-speech server** ; and C Z).= C incoming call logging by the
IVR applications. Two logging tables are provided. Each outgoing message
will be logged in the "Logging...

...ID; an account IID, a message type identifier (e mail, voice mail,
warning, response), and a time stamp. Incoming text-to-speech messages
from the **text -to-speech server** will be logged in the "**Text -to-
speech server** Logging" table of tile database, which is the second
logging table. Each record includes: a user ID, **duration** of the
message (in minutes), a message count, and a time stamp.

The e-mail notification server preferably records a detailed log file.
This file...

22/3,IC,K/2 (Item 2 from file: 349)
DIALOG(R)File 349:PCT Fulltext
(c) 2000 WIPO/MicroPat. All rts. reserv.

00658883

**COMPUTER-BASED PATIENT RECORD AND MESSAGE DELIVERY SYSTEM
SYSTEME DE DOSSIERS INFORMATISES DE PATIENTS ET DE REMISE DE MESSAGES**

Patent Applicant/Assignee:

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Eindhoven , NL

Patent and Priority Information (Country, Number, Date):

Patent: WO 9942932 A2 19990826

Application: WO 99IB192 19990204 (PCT/WO IB9900192)

Priority Application: US 9827125 19980220

Designated States: JP AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: G06F-017/30;

Publication Language: English

Filing Language: English

Fulltext Word Count: 3070

English Abstract

...system. The CPR system further includes a message delivery facility for delivery of textual messages to any of pager, electronic mail, or voice mail (after **text** -to-**speech** synthesis) message delivery vehicles. The **server** system accesses a user specific data store containing speech synthesis profiles which include **prosodic** information of the voices and speech of users, and message delivery profiles which specify which of the aforementioned message delivery vehicles are to be used...

22/3,IC,K/3 (Item 3 from file: 349)

DIALOG(R)File 349:PCT Fulltext

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00592217

A COMMUNICATION SYSTEM ARCHITECTURE

ARCHITECTURE D'UN SYSTEME DE COMMUNICATION

Patent Applicant/Assignee:

MCI COMMUNICATIONS CORPORATION, MCI COMMUNICATIONS CORPORATION , 1133
19th Street, N.W., Washington, DC 20036 , US

EASTEP Guido M

LITZENBERGER Paul R

OREBAUGH Shannon R

ELLIOTT Isaac K

STELLE Rick

SCHRAGE Bruce

BAXTER Craig A

ATKINSON Wesley

KNOSTMAN Chuck

CHEN Bing

VANDERSLUIS Kristan

Inventor(s):

JUN Fang, JUN, Fang , ,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9834391 A2 19980806

Application: WO 98US1868 19980203 (PCT/WO US9801868)

Priority Application: US 97794555 19970203; US 97794114 19970203; US

97794689 19970203; US 97807130 19970210; US 97798208 19970210; US

97795270 19970210; US 97797964 19970210; US 97800243 19970210; US

97798350 19970210; US 97797445 19970210; US 97797360 19970210

Designated States: AU CA GM GW ID JP MX AT BE CH DE DK ES FI FR GB GR IE IT
LU MC NL PT SE

Main International Patent Class: H04M-003/00;

Publication Language: English
Filing Language: English
Fulltext Word Count: 175822

Fulltext Availability:
Detailed Description

Detailed Description

... Unit (ARU) Capabilities 146 1. User Interface 146 L. Message Management 149 1. Multiple Media Message Notification 149 2. Multiple Media Message Manipulation 150 3. **Text** to **Speech** 150 4. Email Forwarding to a Fax Machine 151 5. Pager Notification of Messages Received 151 6. Delivery Confirmation of Voicemail 151 7. Message Prioritization...can be made for directory service as well as for registration (a one time fee plus a monthly fee), call setup, but probably not for **duration** . **Duration** is already charged for the Internet dial in user and is somewhat bundled for the LAN-attached user.

Usage charges for Internet service may be coming soon (as discussed above).

Duration charges are possible for the incoming and outgoing PSTN segments.

Incoming PSTN calls may be charged as the long distance segment by using a special...

?

27/3,IC,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:European Patents
(c) 2000 European Patent Office. All rts. reserv.

00802263

Transaction authorization and alert system
Transaktionsermächtigungs- und -warnsystem
Systeme d'autorisation et d'alarme pour transactions

PATENT ASSIGNEE:

AT&T IPM Corp., (1907680), 2333 Ponce de Leon Boulevard, Coral Gables,
Florida 33134, (US), (applicant designated states: DE;FR;GB)

INVENTOR:

Blonder, Greg E., 112 Mountain Avenue, Summit, New Jersey 07901, (US)
Greenspan, Steven Lloyd, 1566 Ramapo Way, Scotch Plains, New Jersey 07076
, (US)

Mirville, J. Robert, 23 Valley Road, Manalapan, New Jersey 07726, (US)

Sugla, Binay, 161 Van Brackle Road, Aberdeen, New Jersey 07747, (US)

LEGAL REPRESENTATIVE:

Harding, Richard Patrick et al (41295), Marks & Clerk, Nash Court, Oxford
Business Park South, Oxford OX4 2RU, (GB)

PATENT (CC, No, Kind, Date): EP 745961 A2 961204 (Basic)
EP 745961 A3 980715

APPLICATION (CC, No, Date): EP 96303616 960521;

PRIORITY (CC, No, Date): US 455939 950531

DESIGNATED STATES: DE; FR; GB

INTERNATIONAL PATENT CLASS: G07F-007/08; G07F-007/10;

ABSTRACT WORD COUNT: 193

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPAB96	1736
SPEC A	(English)	EPAB96	8640
Total word count - document A			10376
Total word count - document B			0
Total word count - documents A + B			10376

...SPECIFICATION message of FIG. 4 in audio form to the card owner at
telephone set 145, for example. Specifically, IVRS 125 is a processor
that executes **text-to-speech synthesis** programmed **instructions**
designed to use ASCII input, such as one of the messages shown in FIG.
4, to generate a "read aloud" audio rendition of that ASCII...

27/3,IC,K/2 (Item 1 from file: 349)
DIALOG(R)File 349:PCT Fulltext
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00417805

VOICE-OPERATED SERVICES
SERVICES A COMMANDE VOCALE

Patent Applicant/Assignee:

BRITISH TELECOMMUNICATIONS PUBLIC LIMITED COMPANY
ATTWATER David John
WHITTAKER Steven John
SCAHILL Francis James
SIMONS Alison Diane

Inventor(s):

ATTWATER David John
WHITTAKER Steven John
SCAHILL Francis James
SIMONS Alison Diane

Patent and Priority Information (Country, Number, Date):

Patent: WO 9613030 A2-A3 19960502

Application: WO 95GB2524 19951025 (PCT/WO GB9502524)

Priority Application: EP 94307843 19941025

Designated States: AL AM AT AU BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE

HU IS JP KE KG KZ LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU

SD SE SG SI SK TT UA UG US UZ VN KE LS MW SD SZ UG AT BE CH DE DK ES FR

GB GR IE IT LU PT SE BF BJ CF CG CI CM GA GN ML MR NE SN TD TG

Main International Patent Class: G10L-005/06;

Publication Language: English

Fulltext Word Count: 9322

Fulltext Availability:

Detailed Description

Detailed Description

... and one of the recognised town names is tested. If the number is manageable, for example if it is three or fewer, the control means **instructs** (25) the speech **synthesiser** to play an announcement from the message data store 3, followed by recitation of the name, address and telephone number of each entry, generated by the speech synthesiser 1 using **text -to-speech** synthesis, and the process is complete (26). If, on the other hand, the number of entries is excessive then further steps 27, to be discussed...

29/3,IC,K/1 (Item 1 from file: 348)
DIALOG(R)File 348:European Patents
(c) 2000 European Patent Office. All rts. reserv.

00819748

Method and apparatus for modifying voice characteristics of synthesized speech

Verfahren und Vorrichtung zur Veränderung von Stimmeigenschaften synthetisch erzeugter Sprache

Procede et dispositif de modification de caracteristiques de voix pour parole synthetisee

PATENT ASSIGNEE:

AT&T IPM Corp., (1907680), 2333 Ponce de Leon Boulevard, Coral Gables, Florida 33134, (US), (applicant designated states: DE;FR;GB;IT)

INVENTOR:

Buntschuh, Bruce Melvin, 10 Riverbend Road, Berkeley Heights, New Jersey 07922, (US)

LEGAL REPRESENTATIVE:

Watts, Christopher Malcolm Kelway, Dr. (37391), Lucent Technologies (UK) Ltd, 5 Mornington Road, Woodford Green Essex, IG8 0TU, (GB)

PATENT (CC, No, Kind, Date): EP 762384 A2 970312 (Basic)

APPLICATION (CC, No, Date): EP 96306091 960821;

PRIORITY (CC, No, Date): US 522895 950901

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G10L-005/04;

ABSTRACT WORD COUNT: 81

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPAB97	1186
SPEC A	(English)	EPAB97	3745
Total word count - document A			4931
Total word count - document B			0
Total word count - documents A + B			4931

...CLAIMS of speech parameter values are indicative of change to a text to speech synthesizer in corresponding acoustical characteristics of said base synthesized voice;

opening a **text** to **speech synthesizer** with a **command** string containing command-line arguments, wherein said command-line arguments include current present ones of first class speech parameter values;

forming a text string, wherein...

29/3,IC,K/2 (Item 2 from file: 348)
DIALOG(R)File 348:European Patents
(c) 2000 European Patent Office. All rts. reserv.

00736899

Speech synthesis method and apparatus

Verfahren und Vorrichtung zur Sprachsynthese

Methode et dispositif pour la synthese de la parole

PATENT ASSIGNEE:

CANON KABUSHIKI KAISHA, (542361), 30-2, 3-chome, Shimomaruko, Ohta-ku, Tokyo, (JP), (applicant designated states: DE;FR;GB;IT;NL)

INVENTOR:

Otsuka, Mitsuru, c/o Canon K.K., 30-2, 3-chome, Shimomaruko, Ohta-ku, Tokyo, (JP)

Fukada, Toshiaki, c/o Canon K.K., 30-2, 3-chome, Shimomaruko, Ohta-ku, Tokyo, (JP)

Ohora, Yasunori, c/o Canon K.K., 30-2, 3-chome, Shimomaruko, Ohta-ku,
Tokyo, (JP)
Aso, Takashi, c/o Canon K.K., 30-2, 3-chome, Shimomaruko, Ohta-ku, Tokyo,
(JP)

LEGAL REPRESENTATIVE:

Beresford, Keith Denis Lewis et al (28273), BERESFORD & Co. 2-5 Warwick
Court High Holborn, London WC1R 5DJ, (GB)

PATENT (CC, No, Kind, Date): EP 694905 A2 960131 (Basic)

EP 694905 A3 970716

APPLICATION (CC, No, Date): EP 95303570 950525;

PRIORITY (CC, No, Date): JP 94116720 940530

DESIGNATED STATES: DE; FR; GB; IT; NL

INTERNATIONAL PATENT CLASS: G10L-005/04; G10L-007/02;

ABSTRACT WORD COUNT: 162

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
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CLAIMS A	(English)	EPAB96	273
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SPEC A	(English)	EPAB96	14089
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Total word count - document A	14362
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Total word count - document B	0
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Total word count - documents A + B	14362
------------------------------------	-------

...SPECIFICATION of a speech synthesis apparatus used in preferred
embodiments of the present invention.

In FIG. 25, reference numeral 101 represents a keyboard (KB) for
inputting **text** from which **speech** will be **synthesized**, a control
command or the like. The operator can input a desired position on a
display picture surface of a display unit 108 using a pointing device
102...

29/3,IC,K/3 (Item 3 from file: 348)

DIALOG(R)File 348:European Patents

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00602712

**Method and apparatus for scripting a text-to-speech-based multimedia
presentation**

**Verfahren und Gerat zur Steuerung des Arbeitsablaufs einer Umwandlung von
Text in Sprache einer Multimedia-Darstellung**

**Methode et dispositif pour sequencer une presentation multimedia ayant une
conversion texte-parole**

PATENT ASSIGNEE:

Canon Information Systems, Inc., (1553870), 3188 Pullman Street, Costa
Mesa, CA 92626, (US), (applicant designated states: DE;FR;GB;IT)

INVENTOR:

Luther, Willis J., 5 Spicewood Way, Irvine, California 92715, (US)

LEGAL REPRESENTATIVE:

Beresford, Keith Denis Lewis et al (28273), BERESFORD & Co. 2-5 Warwick
Court High Holborn, London WC1R 5DJ, (GB)

PATENT (CC, No, Kind, Date): EP 598597 A1 940525 (Basic)

EP 598597 B1 990224

APPLICATION (CC, No, Date): EP 93309146 931116;

PRIORITY (CC, No, Date): US 978336 921118

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G06F-003/16;

ABSTRACT WORD COUNT: 147

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
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CLAIMS B (English)	9907	645
CLAIMS B (German)	9907	547
CLAIMS B (French)	9907	787
SPEC B (English)	9907	3256
Total word count - document A		0
Total word count - document B		5235
Total word count - documents A + B		5235

...SPECIFICATION in script buffer 31 is provided to processor 32 which separates the text narration from the multimedia commands and provides the text narration to the **text -to-speech** converter 26. The presence of multimedia commands is detected by an action token detector 34 which identifies the beginning of each scripting command. The action...

...intended. In general, the multimedia scripting command can include commands to incorporate further text files 36 and feed the text from those text files to **text -to-speech** converter 26, commands to obtain MIDI files 37 and feed the MIDI music in those files to MIDI **synthesizer** 28, **commands** to obtain bit map image files 39 and to feed the still video information in those bit map image files to video monitor 17, commands...

29/3,IC,K/4 (Item 4 from file: 348)

DIALOG(R)File 348:European Patents

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00596781

Speech recognition interface system suitable for window systems and speech mail systems

Spracherkennungs-Schnittstellensystem, das als Fenstersystem und Sprach-Postsystem verwendbar ist

Systeme d'interface de reconnaissance de la parole adapte pour des systemes a fenetre et systemes de messagerie a parole

PATENT ASSIGNEE:

KABUSHIKI KAISHA TOSHIBA, (213130), 72, Horikawa-cho, Saiwai-ku, Kawasaki-shi, Kanagawa-ken 210-8572, (JP), (Proprietor designated states: all)

TOSHIBA SOFT ENGINEERING COMPANY LIMITED, (1732010), 1385 Shin-cho, Oume-shi, Tokyo, (JP), (Proprietor designated states: all)

INVENTOR:

Hashimoto, Hideki, 502 Fulola-Miyazkidai, 1378 Miginu, Miyamae-ku, Kawasaki-shi, Kanagawa-ken, (JP)

Nagata, Yoshifumi, TOSHIBA-Kikuna-ryo A424, 217 Mamedo, Kouhoku-ku, Yokohama-shi, Kanagawa-ken, (JP)

Seto, Shigenobu, 4-24-7, Kishiya, Tsurumi-ku, Yokohama-shi, Kanagawa-ken, (JP)

Takebayashi, Yoichi, 1660-A105, Komaoka-cho, Tsurumi-ku, Yokohama-shi, Kanagawa-ken, (JP)

Shinchi, Hideaki, Fulola-Miyazakidai, 1378, Maginu, Miyamae-ku, Kawasaki-shi, Kanagawa-ken, (JP)

Yamaguchi, Koji, 3-16-46, Fujimi, Urayasu-shi, Chiba-ken, (JP)

LEGAL REPRESENTATIVE:

Lehn, Werner, Dipl.-Ing. et al (7474), Hoffmann Eitle, Patent- und Rechtsanwalte, Arabellastrasse 4, 81925 Munchen, (DE)

PATENT (CC, No, Kind, Date): EP 607615 A1 940727 (Basic)

EP 607615 B1 990915

APPLICATION (CC, No, Date): EP 93121031 931228;

PRIORITY (CC, No, Date): JP 92358597 921228; JP 9378920 930312; JP 93256405 930920

DESIGNATED STATES: DE; FR

INTERNATIONAL PATENT CLASS: G06F-003/16

ABSTRACT WORD COUNT: 179

NOTE:

Figure number on first page: 6

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9937	2642
CLAIMS B	(German)	9937	2250
CLAIMS B	(French)	9937	3193
SPEC B	(English)	9937	43846
Total word count - document A			0
Total word count - document B			51931
Total word count - documents A + B			51931

...SPECIFICATION response pair. Here, the operation may not necessarily be the speech controlled one. Also, the response is described as a command, where "synth()" is a **command** to output the **synthesized** speech with its argument as the **text** of the **speech** output, and "play()" is a command to output the data specified by its argument as the waveform data. Here, \$<cat> in the argument...

29/3,IC,K/5 (Item 5 from file: 348)

DIALOG(R)File 348:European Patents

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00462745

Method and apparatus for segmental unit representation in text-to-speech synthesis.

Verfahren und Einrichtung zur Darstellung von Segmenteinheiten zur Text-Sprache-Umsetzung.

Methode et dispositif pour la representation d'unites segmentaires pour la conversion texte-parole.

PATENT ASSIGNEE:

International Business Machines Corporation, (200120), Old Orchard Road, Armonk, N.Y. 10504, (US), (applicant designated states: DE;FR;GB)

IBM SEMEA S.r.l., (1179640), Via Fara, 35, P.O. Box 137, I-20124 Milan, (IT), (applicant designated states: IT)

INVENTOR:

Giustiniani, Massimo, Via Carlo Fadda 19, I-00173 Rome, (IT)

Pierucci, Piero, Via P. Mengoli 14, I-00146 Rome, (IT)

LEGAL REPRESENTATIVE:

Lettieri, Fabrizio (59683), IBM SEMEA S.p.A., Direzione Brevetti, MI SEG 534, P.O. Box 137 P.O. Box 137, I-20090 Segrate (Milano), (IT)

PATENT (CC, No, Kind, Date): EP 515709 A1 921202 (Basic)

APPLICATION (CC, No, Date): EP 91108575 910527;

PRIORITY (CC, No, Date): EP 91108575 910527

DESIGNATED STATES: DE; FR; GB; IT

INTERNATIONAL PATENT CLASS: G10L-005/04;

ABSTRACT WORD COUNT: 132

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPABF1	753
SPEC A	(English)	EPABF1	5505
Total word count - document A			6258
Total word count - document B			0
Total word count - documents A + B			6258

...CLAIMS the determination of the speech feature vectors is obtained

taking the feature vectors of said AEHMM corresponding to the most probable labels.

6. A concatenative **text -to-speech** synthesizer system including a Text Input means (100) for entering text to be synthesized, a Text Processor (101) for converting the graphemic input into a...

...feature vectors for said Synthesis Filter (106) and a Back-Transformation Processor (SU14) which transforms the domain of spectral coefficient representation in order to be **directly** used by said **Synthesis** Filter (106).

7. The **text -to-speech** synthesizer system of claim 6 in which said Segmental Unit Linker (105) includes a Stretch by Copy Processor (SU21) producing a sequence of labels with...

...to phonotactical constraints of the language and a Coefficients Back-Transformation Processor (SU25) which transforms the domain of spectral coefficient representation in order to be **directly** used by said **Synthesis** Filter (106).

8. The text-to speech synthesizer system of claim 7 wherein said optimality criterion used in said AEHMM Interpolation Processor (SU24) consists in...

29/3,IC,K/6 (Item 1 from file: 349)

DIALOG(R)File 349:PCT Fulltext

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00665283

METHOD AND APPARATUS FOR PERFORMING HANDSFREE OPERATIONS AND VOICING TEXT WITH A CDMA TELEPHONE

PROCEDE ET APPAREIL POUR UNE UTILISATION MAINS LIBRES ET UNE TRANSMISSION DE TEXTE PAR LA VOIX AVEC UN TELEPHONE AMCR

Patent Applicant/Assignee:

QUALCOMM INCORPORATED, QUALCOMM INCORPORATED , 6455 Lusk Boulevard, San Diego, CA 92121 , US

Inventor(s):

MOHANTY Bibhu, MOHANTY, Bibhu , 4028 Mahaila Avenue &C, San Diego, CA 92122 , US

SORENSEN Cristian, SORENSEN, Cristian , 445 Delage Drive, Encinitas, CA 92024 , US

Patent and Priority Information (Country, Number, Date):

Patent: WO 9949681 A1 19990930

Application: WO 99US6360 19990323 (PCT/WO US9906360)

Priority Application: US 9879406 19980325; US 98182582 19981029

Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES

FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU

LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA

UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ UG ZW AM AZ BY KG KZ MD RU TJ

TM AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ CF CG CI

CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04Q-007/32;

International Patent Class: H04M-003/50;

Publication Language: English

Filing Language: English

Fulltext Word Count: 5372

Fulltext Availability:

Detailed Description

Detailed Description

... and affect the safety of the driver as well as those around the

driver.

The art of speech synthesis has seen many improvements, and today **text** to **speech** converters are commercially available. Cellular telephones which supply audio feedback are known ...feedback of numbers in response to keys pressed on a keypad. Another example is U.S. Patent No. 5,095,503, entitled "CELLULAR TELEPHONE CONTROLLERWITH **SYNTHESIZED** VOICE FEEDBACK FOR **DIRECTORY** NUMBER CONFIRMATION AND CALL STATUS". Speech recognition is another area that now offers commercial solutions to those wishing to employ voice commands in a system...

29/3,IC,K/7 (Item 2 from file: 349)
DIALOG(R) File 349:PCT Fulltext
(c) 2000 WIPO/MicroPat. All rts. reserv.

00365605

METHOD AND APPARATUS FOR MULTIFACETED ELECTROENCEPHALOGRAPHIC RESPONSE ANALYSIS (MERA)

PROCEDE ET APPAREIL D'ANALYSE DE REPONSES ELECTROENCEPHALOGRAPHIQUES A FACETTES MULTIPLES

Patent Applicant/Assignee:

FARWELL Lawrence Ashley

Inventor(s):

FARWELL Lawrence Ashley

Patent and Priority Information (Country, Number, Date):

Patent: WO 9426162 A1 19941124

Application: WO 94US4851 19940503 (PCT/WO US9404851)

Priority Application: US 9357607 19930505

Designated States: AU BR CA CN JP KR RU AT BE CH DE DK ES FR GB GR IE IT LU
MC NL PT SE

Main International Patent Class: A61B-005/04;

Publication Language: English

Fulltext Word Count: 15100

Fulltext Availability:

Claims

Claim

... PC could send a command and deassert the task bit lines before the robot (which runs very slowly) has had a chance to buffer the **command** .

The speech **synthesizer** system 190 from DEC includes a board for the PC and a speaker to place on the robot (DECtalk PC **text** -to-**speech** system from Digital Equipment Corporation). Connections were made with a standard

File 350:Derwent WPIX 1963-2000/UD,UM &UP=200046
(c) 2000 Derwent Info Ltd
File 347:JAPIO Oct 1976-2000/May(UPDATED 000915)
(c) 2000 JPO & JAPIO
File 344:Chinese Patents ABS Apr 1985-2000/Aug
(c) 2000 European Patent Office
File 348:European Patents 1978-2000/Sep W04
(c) 2000 European Patent Office
File 349:PCT Fulltext 1983-2000/UB=20000921, UT=20000908
(c) 2000 WIPO/MicroPat

Set	Items	Description
S1	402	AU=(SIMPSON D? OR CURRY J? OR MCALLISTER A?)
S2	5	S1 AND (SPEECH AND MESSAGE?)

?t /3,ic,k/1-5

2/3,IC,K/1 (Item 1 from file: 350)

DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012865366
WPI Acc No: 2000-037199/200003
Related WPI Acc No: 2000-246226; 2000-450673
XRPX Acc No: N00-027910

Personal dial tone service of intelligent telephone network

Patent Assignee: BELL ATLANTIC NETWORK SERVICES (BELL-N)

Inventor: FARRIS R D; **MCALLISTER A I** ; STRAUSS M J

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5978450	A	19991102	US 97828959	A	19970328	200003 B

Priority Applications (No Type Date): US 97828959 A 19970328

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5978450	A	20	H04M-007/00	

International Patent Class (Main): H04M-007/00

...Inventor: **MCALLISTER A I**

Abstract (Basic):

... A service control point comprises a database of call processing records, for controlling several services along with the central office. Signaling **messages** are communicated between the service control point and the central office. An INDEPENDENT CLAIM is also included for personalized telecommunication services processing method ...

...The service uses **speech** based identification, thereby eliminating the burden on the subscriber to dial in long strings of identifying digits ...

2/3,IC,K/2 (Item 2 from file: 350)

DIALOG(R)File 350:Derwent WPIX
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011341453
WPI Acc No: 1997-319358/199729
Related WPI Acc No: 1994-218202
XRPX Acc No: N97-264409

Automated subscriber telephone number providing method - prompting user to speak name and location of sought party, and digitising responses before feeding them to speech recognition devices, whose outputs are used to search database for corresponding number

Patent Assignee: BELL ATLANTIC NETWORK SERVICES (BELL-N)

Inventor: CASEY K M; **CURRY J E** ; HANLE J P; HAYDEN J B; **MCALLISTER A I** ; MEADOR F E; TRESSLER R C

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5638425	A	19970610	US 92992207	A	19921217	199729 B
			US 94333988	A	19941102	

Priority Applications (No Type Date): US 94333988 A 19941102; US 92992207 A 19921217

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5638425	A	29	H04M-001/64	CIP of application US 92992207

International Patent Class (Main): H04M-001/64

International Patent Class (Additional): G10L-005/00

... **prompting user to speak name and location of sought party, and digitising responses before feeding them to speech recognition devices, whose outputs are used to search database for corresponding number**
...Inventor: **CURRY J E** ...

...**MCALLISTER A I**

...Abstract (Basic): a telephone user to an automated directory assistance station, upon a user dialling a predetermined number on a telephone. The user responds to a stored **message** , by speaking a name of a location of a sought subscriber. A second stored **message** prompts the user to speak the last name of the sought subscriber. The responses from the user are encoded into first and second digital signals which are compatible with two **speech** recognition devices. The signals are transmitted to the **speech** recognition devices which use word recognition and phoneme recognition, respectively...

...The output signals from the **speech** recognition devices are decoded and a probability level signal is associated with each decoded signal. The probability level signals are combined according to a predetermined...

...signals, associated with the highest probability level are selected. The second selected signal is used to obtain a corresponding directory number from a database. A **message** is transmitted to the user, articulating the directory number...

...USE/ADVANTAGE - E.g. for automatic processing of directory assistance calls in telecommunication network. Uses available **speech** recognition equipment in unique manner, to attain improved level of effectiveness. Minimises necessity to rely on operator intervention. Maximises successful provision of required assistance...

...Title Terms: **SPEECH** ;

2/3,IC,K/3 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

(c) 2000 Derwent Info Ltd. All rts. reserv.

009950489

WPI Acc No: 1994-218202/199426

Related WPI Acc No: 1997-319358

XRPX Acc No: N94-172289

Providing subscriber telephone numbers to telephone users - using speech recognition to decode area and name prompted from user and articulates corresp. code and number retrieved from database

Patent Assignee: BELL ATLANTIC NETWORK SERVICES (BELL-N)

Inventor: CASEY K M; **CURRY J E** ; HANLE J P; HAYDEN J B; **MCALLISTER A I** ;

MEADOR F E; TRESSLER R C; **MCALLISTER A**

Number of Countries: 045 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9414270	A1	19940623	WO 93US12247	A	19931216	199426 B
AU 9458033	A	19940704	AU 9458033	A	19931216	199437

Priority Applications (No Type Date): US 92992207 A 19921217

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
-----------	------	--------	----------	--------------

WO 9414270	A1	E 46	H04M-001/64	
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Designated States (National): AT AU BB BG BR BY CA CH CZ DE DK ES FI GB

HU JP KP KR KZ LK LU LV MG MN MW NL NO NZ PL PT RO RU SD SE SK UA UZ VN

Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LU MC NL

OA PT SE

AU 9458033 A H04M-001/64 Based on patent WO 9414270

International Patent Class (Main): H04M-001/64

International Patent Class (Additional): G10L-005/00; G10L-005/06;

G10L-007/00; G10L-007/08; G10L-009/00; G10L-009/06; H04M-003/42

... using speech recognition to decode area and name prompted from user and articulates corresp. code and number retrieved from database

...Inventor: **CURRY J E** ...

...**MCALLISTER A I** ...

...**MCALLISTER A**

...Abstract (Basic): The method involves enabling automated station to respond to a set dialled number to prompt a caller by a recorded **message** to give a desired location. The response is digitised and simultaneously input to word and phoneme recognition devices, which each output a translation signal and...

...ADVANTAGE - Efficient. Acceptable and pleasing to user. Uses available **speech** recognition devices. Need for operator intervention minimised

...

...Title Terms: **SPEECH** ;

2/3,IC,K/4 (Item 1 from file: 349)

DIALOG(R)File 349:PCT Fulltext

(c) 2000 WIPO/MicroPat. All rts. reserv.

00353482

MECHANIZED DIRECTORY ASSISTANCE

SERVICE DE RENSEIGNEMENTS TELEPHONIQUES MECANISE

Patent Applicant/Assignee:

BELL ATLANTIC NETWORK SERVICES INC

Inventor(s):

MEADOR Frank E

CASEY Kathleen M

CURRY James E

McALLISTER Alexander I

TRESSLER Robert C

HAYDEN James B

HANLE John P

Patent and Priority Information (Country, Number, Date):

Patent: WO 9414270 A1 19940623

Application: WO 93US12247 19931216 (PCT/WO US9312247)

Priority Application: US 92992207 19921217

Designated States: AT AU BB BG BR BY CA CH CZ DE DK ES FI GB HU JP KP KR KZ

LK LU LV MG MN NO NZ PL PT RO RU SD SE SK UA UZ VN AT BE CH DE DK ES FR

GB GR IE IT LU PT SE BF BJ CF CG CI CM GA GN ML MR NE SN TD TG

Main International Patent Class: H04M-001/64;

International Patent Class: H04M-003/42; G10L-005/00; G10L-007/00;

G10L-009/00; G10L-005/06; G10L-007/08; G10L-009/06;

Publication Language: English

Fulltext Word Count: 6709

Inventor(s):

MEADOR Frank E

CASEY Kathleen M

CURRY James E

McALLISTER Alexander I

TRESSLER Robert C

HAYDEN James B

HANLE John P

Fulltext Availability:

Detailed Description

Claims

English Abstract

An automated directory assistance system with voice processing unit for use in a telecommunication network includes multiple **speech** recognition devices comprising a word recognition device, a phoneme recognition device, and an alphabet recognition device. A caller is prompted to speak the city or...

Detailed Description

... connect the telephone to an audio digital interface system and causing a first prestored prompt to be provided to the user from system memory.

This **message** instructs the user to spell letter-by letter the last name of the subscriber whose telephone number is desired. Each time a letter of the...

...the number of such matches in reading through the entire main memory to be stored in a match counter.

A selected one of three recorded **messages** is then transmitted to the user with the selected **message** corresponding to one of four different situations.

2/3,IC,K/5 (Item 2 from file: 349)
DIALOG(R)File 349:PCT Fulltext
(c) 2000 WIPO/MicroPat. All rts. reserv.

00318332

METHOD AND SYSTEM FOR HOME INCARCERATION

PROCEDE ET SYSTEME PERMETTANT L'INCARCERATION A DOMICILE

Patent Applicant/Assignee:

BELL ATLANTIC NETWORK SERVICES INC

Inventor(s):

D'ALESSIO Frederick D

WEGLEITNER Mark A

MCALLISTER Alexander I

KEOPPE Alfred C

Patent and Priority Information (Country, Number, Date):

Patent: WO 9305605 A1 19930318

Application: WO 92US7645 19920911 (PCT/WO US9207645)

Priority Application: US 91758051 19910912

Designated States: AT AU BB BG BR CA CH CS DE DK ES FI GB HU JP KP KR LK LU

MG MN MW NL NO RU SD SE AT BE CH DE DK ES FR GB GR IE IT LU MC NL SE BF

BJ CF CG CI CM ML MR SN TD TG

Main International Patent Class: H04M-011/04;

Publication Language: English

Fulltext Word Count: 4547

Inventor(s):

D'ALESSIO Frederick D

WEGLEITNER Mark A

MCALLISTER Alexander I

KEOPPE Alfred C

Fulltext Availability:

Detailed Description

Claims

English Abstract

...telephone network including a telephone (58) on the premises of the location of confinement and a control center (48). Voice verification, using voice analysis of **speech** transmitted in a telephone call from the site (58) to the center (48) is performed during periodic testing. A voice template vocabulary is established for...

Detailed Description

... premises by communicating with the individual via a telephone network, identifying the location by utilizing caller line identification and identifying the individual by voice identification **speech** processing.

Backaround Art

Theconcept of home incarceration has evolved as an alternative to detention in government jail and prison facilities. In cases of relatively light...individual to be verlf ied. Such identification attempts likely would not be successful if the system serves a large number of detainees or if the **speech** of the called party is slurred by the influence of drug or alcohol abuse. Enforcement personnel frequently must be dispatched *to the confinement sites to...377 contemplates the use of a voiceprint as a means for remote identification of a prisoner. Audio spectral analysis is performed and -lz applied to **speech** transmitted over a telephone line to determine a match with a probationer's voiceprint.

File 647: CMP Computer Fulltext 1988-2000/Sep W1
(c) 2000 CMP
File 275: Gale Group Computer DB(TM) 1983-2000/Sep 27
(c) 2000 The Gale Group
File 674: Computer News Fulltext 1989-2000/Sep W1
(c) 2000 IDG Communications
File 98: General Sci Abs/Full-Text 1984-2000/Aug
(c) 2000 The HW Wilson Co.
File 624: McGraw-Hill Publications 1985-2000/Sep 21
(c) 2000 McGraw-Hill Co. Inc
File 636: Gale Group Newsletter DB(TM) 1987-2000/Sep 27
(c) 2000 The Gale Group
File 148: Gale Group Trade & Industry DB 1976-2000/Sep 27
(c) 2000 The Gale Group
File 696: DIALOG Telecom. Newsletters 1995-2000/Sep 26
(c) 2000 The Dialog Corp.

Set	Items	Description
S1	16090	TEXT? ? (2W) (SOUND OR AUDIO? OR VOICE? OR SPEECH)
S2	10855	(SPEECH OR VOICE) (2N) (SYNTHE? OR GENERAT?)
S3	25289	S1 OR S2
S4	3630557	(WEB OR NETWORK OR W3 OR INTERNET OR INTRANET OR SERVER? OR SITE? OR WEB() PAGE?)
S5	8033	S3(S)S4
S6	1374476	(SYNTHE? OR GENERAT?)
S7	93203	(ROUT? OR DELIVER? OR SEND OR SENT OR TRANSMIT? OR TRANSMIS? OR PASS? OR REMIT? OR DOWNLOAD)(5N) (AUDIO OR SPEECH OR VOICE OR SOUND)
S8	3747081	USER? OR CUSTOMER? OR CLIENT? OR SUBSCRIB?
S9	191341	(PITCH? OR DURATION OR APTITUDE OR (ATTACK OR DECAY)(2N) ENVELOP? OR (SYNTHE? () INSTRUCT?) OR CONTROL? (2N) PARAMET?)
S10	12038	(WAVEFORM? OR WAVE()FORM? ?)
S11	140	DIGIT?()SEQUENC?
S12	5537	CONCATENAT? OR (SPEECH? OR SOUND? OR VOICE)(2N)(FRAGMENT? - OR SAMPL?)
S13	21047	(PROSOD? OR ACCENTUAT? OR INTONATION?) OR (NATURAL OR - HIGH()QUALITY)(3N) (SOUND? OR SPEECH OR VOICE)
S14	843	S5(S)S7
S15	5	S14(S)S9
S16	28	S5(S)S9 NOT S15
S17	9	S16 NOT PY>1997
S18	8	RD (unique items)
S19	157	S5(S)S8(S)(S9:S13)
S20	24	S19(S)SYNTHE?SIZ?
S21	7	S20 NOT PY>1997
S22	5	RD (unique items)

full text
NPL
RA

15/3,K/3 (Item 2 from file: 275)
DIALOG(R) File 275:Gale Group Computer DB(TM)
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01235466 SUPPLIER NUMBER: 07142699

An efficient multiplexing technique for packet-switched voice-data networks.

Choi, J.K.; Un, C.K.

Proceedings of the IEEE, v76, n9, p1254(3)
Sept, 1988

ISSN: 0018-9219

LANGUAGE: ENGLISH

RECORD TYPE: ABSTRACT

ABSTRACT: A new computationally-efficient packet-switched voice-data **network** multiplexing technique uses a synchronous frame structure with the same **duration** as the **voice** packet **generation** interval to enable the synchronous **transmission** of **voice** packets without loss or clipping. Packets are sequenced through the use of inter-arrival--delay--times and sequence numbers. The intervals and active voice periods...

...derived from the received packet streams through use of the frame format. The frame structure is applicable to a variety of enhanced services with varying **voice** **transmission** rates and packet sizes.

15/3,K/4 (Item 1 from file: 674)
DIALOG(R) File 674:Computer News Fulltext
(c) 2000 IDG Communications. All rts. reserv.

082505

Gearhead - Speechifyin' software

Byline: Mark Gibbs

Journal: Network World Page Number: 46

Publication Date: March 27, 2000

Word Count: 494 Line Count: 44

Text:

... 28, page 48), we discussed a cool utility named Talking Stocks from 4Developers (www.4developers.com). Gearhead has been intrigued by software that talks since **speech** - **generating** chips became available about 15 years ago. We remember well those distorted robot voices that sounded like a tourist from Eastern Europe with a bad...

...WillowTalk has a range of predefined voices that imitate male and female tonality quite well. You can also define your own voices in terms of **pitch**, speed and volume, and the product allows for custom dictionaries so you can define the pronunciation of special words. The reading scripts feature is odd, to say the least: You fill in a grid with the voice you want in one column and the **text** for that **voice** in the other and the voice reads the script. One of these days Gearhead plans to create a completely synthesized reading of MacBeth (<http://sailor...>

... speech to a file that lets you include synthesized voices in other applications. Another fun speech utility is SayIt from AnalogX (www.analogx.com/contents/download/audio/sayit.htm). AnalogX has a lot of public domain software for Windows on its **Web site**, including something called SayIt. SayIt is simple and was designed along the lines of Speak 'n Spell. It has a text entry window where you can enter up to 500 text characters and four sliders that let you change **pitch**, speed, modulation and cascade. (AnalogX omits explaining what these last two attributes actually do - get 'em wrong and the voice can sound awful.) You can simply have the text read to you or you can save the **synthesized**

15/3,K/1 (Item 1 from file: 647)
DIALOG(R)File 647:CMP Computer Fulltext
(c) 2000 CMP. All rts. reserv.

01124894 CMP ACCESSION NUMBER: CWK19970505S0042
Street Technologies Paves Way For Sound (Intranet Watch)
John Evan Frook
COMMUNICATIONSWEEK, 1997, n 661, PG8
PUBLICATION DATE: 970505
JOURNAL CODE: CWK LANGUAGE: English
RECORD TYPE: Fulltext
SECTION HEADING: Top of the News
WORD COUNT: 173

TEXT:

Corporate buyers have saved millions purchasing computers devoid of sound cards, but now those unhearing machines are useless for **delivery** of **audio** training materials over corporate intranets. That's the issue Street Technologies Inc., White Plains, N.Y., is tackling with its StreetSound parallel port sound card...

...sound. Street Technologies CEO Stephen Gott said the card was developed to help support efforts of Street's sister company, Learning Tree International, which develops **text**, **audio** and video computer-based training programs at www.learningtree.com. After **pitching** 50 different CIOs on developing multimedia training materials for intranets, Gott said he found that 90 percent of their installed seats had no sound. Street...

...95 sound card monthly and this week is launching a multimedia help desk for StreetSound, accessed via www.streetinc.com, to demonstrate the power of **Web** -training tools.

15/3,K/2 (Item 1 from file: 275)
DIALOG(R)File 275:Gale Group Computer DB(TM)
(c) 2000 The Gale Group. All rts. reserv.

02074428 SUPPLIER NUMBER: 19520438 (USE FORMAT 7 OR 9 FOR FULL TEXT)
On the Web, voices carry; high quality and low bandwidth. (Voxware)
(Company Business and Marketing)
PC Magazine, v16, nSpeiss, p12(1)
Summer, 1997
ISSN: 0888-8507 LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 266 LINE COUNT: 00024

... with the goal of making Internet telephony a competitive alternative to today's telephone network.

Voxware's core technology creates a mathematical model of human **speech** that can then be efficiently **delivered** over the **Internet**. Once the software models the data, the voice can do all kinds of gymnastics, altering **pitch**, speed, resonance, and other characteristics. In one application of this technology, an entertainment company could model an actor's voice for a cartoon character, ensuring that the character--complete with computer-generated **voice**--could outlive the actor. Spooky.

The ability to transform a human voice and then send it over the Web raises a whole new area of...

voice to disk. Some of the voices Gearhead got out of SayIt were great - clear and easily understood. These speaking programs are great fun and can be used to **generate speech** for other application programs or **Web sites**. Ein, zwei, drei, vier. Synthesis to gh@gibbs.com.

15/3,K/5 (Item 1 from file: 148)
DIALOG(R)File 148:Gale Group Trade & Industry DB
(c)2000 The Gale Group. All rts. reserv.

11557320 SUPPLIER NUMBER: 58079713 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Fastcomm Introduces Enhanced Feature Set for the Voice Over Packet

Marketplace.

Business Wire, 1331

Dec 8, 1999

LANGUAGE: English RECORD TYPE: Fulltext

WORD COUNT: 925 LINE COUNT: 00081

... enhance our reputation with current and potential customers."

Integrated Call Detail Recording

The Integrated Call Detail Recording (iCDR) enhancement captures call activity details from each **site** on the **network**. After a call is terminated, the MetroLAN(TM) or GlobalStack(TM) packet **voice router** will **generate** a message that contains the call details. The captured data includes calling and called parties, call **duration** (to the second), call routing information (whether routed over FR or IP), and disconnection reason.

The iCDR message is routed as an IP packet through...

?

18/3,K/1 (Item 1 from file: 275)

DIALOG(R)File 275:Gale Group Computer DB(TM)

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01549059 SUPPLIER NUMBER: 12972691 (USE FORMAT 7 OR 9 FOR FULL TEXT)

Planning for 1995: the future is now. (technology strategies for the future) (overview of four articles on strategic technology planning) (Special Report: 1995)

Battelle, John; Eliot, Lance B.; Rothfelder, Jeffrey; Steinberg, Don
Corporate Computing, v1, n6, p166(15)
Dec, 1992

ISSN: 1065-8610

LANGUAGE: ENGLISH

RECORD TYPE: FULLTEXT

WORD COUNT: 7255

LINE COUNT: 00564

... Ashton-Tate will bring to light the industry's best new acronym: BLObs. You can find BLObs (binary large objects; compound objects that can contain **text**, graphics, **sound**, video, and other information) in InterBase, the jewel in Ashton-Tate's software portfolio. InterBase is a Unix-based relational database **server** engine that supports both SQL and its own data-manipulation language. Its proprietary language extends the standard relational model significantly: it's geared for high...

...bursts of incoming information such as a sales transaction. Yet it also supports the kind of database access typical to a desktop PC user: long-**duration** browsing, editing, and printing of records. Borland calls this odd combination on-line complex processing, in which real-time transactions can be posted while users...

18/3,K/2 (Item 2 from file: 275)

DIALOG(R)File 275:Gale Group Computer DB(TM)

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01548151 SUPPLIER NUMBER: 13229548 (USE FORMAT 7 OR 9 FOR FULL TEXT)

Microsoft goes for hard sell. (Microsoft's marketing strategy for its new Access database management system) (PC User News) (Brief Article)

PC User, n198, p17(1)

Nov 18, 1992

DOCUMENT TYPE: Brief Article

ISSN: 0263-5720

LANGUAGE: ENGLISH

RECORD TYPE: FULLTEXT

WORD COUNT: 300

LINE COUNT: 00023

Microsoft will be staging road-shows in London, Birmingham, Manchester, Edinburgh, Bath and Dublin. Some dealers will be holding their own events.

Microsoft is **pitching** Access at users with little or no programming skills, enabling them to build databases with **text**, numbers, **sound** and full-motion video. Its QGBE graphical query tool can also analyse data between dBase, Paradox, Btrieve and Microsoft SQL **Server** formats.

Mike Farrow, a consultant developer with a beta site for Access, Channel Business Systems, believes there will be a large market for the product...

18/3,K/3 (Item 1 from file: 636)

DIALOG(R)File 636:Gale Group Newsletter DB(TM)

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03467068 Supplier Number: 47147989 (USE FORMAT 7 FOR FULLTEXT)

-IBM: IBM and Eloquent Technology Inc. Announce speech technology alliance
M2 Presswire, pN/A

Feb 24, 1997

Language: English Record Type: Fulltext

Document Type: Newswire; Trade

Word Count: 651

(USE FORMAT 7 FOR FULLTEXT)

TEXT:

...IBM and Eloquent Technology Inc. Announce speech technology alliance (C)1994-97 M2 COMMUNICATIONS LTD RDATE:210297 * IBM to license rights to ETI's advanced **text -to-speech** system IBM and Eloquent Technology Inc. (ETI) have recently announced in the US that IBM has acquired certain exclusive rights to Eloquent's powerful **text -to-speech** technology system. As part of the agreement, IBM and Eloquent will work closely to integrate **text -to-speech** functions into future IBM products and applications that are part of the IBM VoiceType family. ETI will continue to license and support its toolkit product...

...to enhance the consumer's experience by extending speech technology to applications, products and appliances of all shapes and sizes." ETI-Eloquence is a flexible **text -to-speech** system that produces high-quality speech with natural sounding intonation. The ETI- Eloquence system provides nine built-in voices, including those of adults and children, both male and female. Developers and end-users can easily create additional voices by **controlling** such **parameters** as gender, breathiness, roughness, **pitch** fluctuation and speaking rate. The linguistic models and specialised development tools underlying Eloquence make it highly extensible and customizable. In addition, the technology provides a robust development platform that both IBM and ETI plan to exploit as the market for high quality **text -to-speech** solutions continues to develop. "We are very pleased that IBM recognised the potential of our technology," said Sue Hertz, Ph.D., president of Eloquent Technology...

...a broad variety of speech-enabled products, and will provide users with access to many interactive applications that take advantage of a combined speech-to-**text** /**text -to-speech** product and toolkit." Notes to editor Eloquent Technology, Inc. ETI, located in Ithaca, New York, was founded in 1983 by Sue Hertz, Ph.D., explicitly for the purpose of developing and marketing **text -to-speech** software. ETI has been the recipient of numerous government grants and contracts for **text -to-speech** research and development. The first version of Eloquence was released in early 1995. ETI-Eloquence is suitable for a wide range of applications, which include reading and speaking aids, CD-ROM edutainment products, telephony and integrated voice response applications, **Internet** talking pages, information and warning systems, and many others. For more information on ETI or its products, call 00 1 607-266- 7025. **Internet** users can access the ETI home page on the World Wide **Web** at <http://www.eloq.com> A UK English version of ETI-Eloquence will be available later this year. IBM Speech Systems IBM, with a family...

...3.0 for Windows 95. * Product and service names are the trademarks or registered trademarks of International Business Machines Corporation, or their respective owners. For **Internet** users, IBM offers complete information about the company, its products, services and technology on the World Wide **Web** . The IBM VoiceType home page is at www.software.ibm.com/is/voicetype. CONTACT: James Lloyd, Charles Barker Tel: +44 (0)171 830 8493 Fax...

18/3,K/4 (Item 2 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)

(c) 2000 The Gale Group. All rts. reserv.

02683352 Supplier Number: 45442536 (USE FORMAT 7 FOR FULLTEXT)
EDGE OF CHAOS: Current Perspectives on Interactive Advertising Paul Kagan
Conference on Interactive Advertising
Multimedia & Videodisc Monitor, v13, n4, pN/A
April, 1995
Language: English Record Type: Fulltext
Document Type: Newsletter; Trade
Word Count: 2861

(USE FORMAT 7 FOR FULLTEXT)

TEXT:

...get the lines in." He called interactivity a marketing discipline -- as opposed to an advertising or promotional discipline -- and offered the example of a Godiva **Internet site** that informs about the "lusciousness of chocolate" and also includes an online candy store. Hauptschein commented that interactivity must be thought of as a content...

...3400, 30 South Wacker Drive, Chicago IL 60606, 312/750-5000). * Marty Levin (vice president, Microsoft Advanced Technology Division; creative director of the pending Microsoft **Network**) described the current online services market as the "first step up the bandwidth scale," with communications being the current killer application. Regarding Microsoft's business model, he indicated that on Microsoft **Network**, the information providers (as opposed to the service operator) will be making the lion's share of the revenue. Levin said, "Today we have connectivity...t see much of it in informercials." He said that when talent performs, much more merchandise is sold than when the person "gets too involved **pitching** the product." Paxton reminded the audience that a telethon (which is long program for charity) raises the most money when the performers are on stage. Paxton said, "It's time for advertisers to get back to sponsoring shows -- not just **pitching** products." He reminded attendees of the day when the Texaco Television Theater, hosted by Sid Caesar, "had the Texaco Star emblem on screen for over forty minutes of programming time." He said, "There is tremendous room for diversification in infomercials," which today **pitch** five things: "thinness, muscles, hair, psychics, and finding a mate." Supporting Tom Grieb's statement about how interactivity assists in local markets, Paxton told the...

...to "old programmers to create new content." He also admitted that interactive tools currently "stink." He predicted that people will soon get burned on the **Internet**, as an average **site** can handle only three to twelve simultaneous callers. He compared the cruise ship industry to the online services industry. In surveys of potential cruise ship...
...it again. Leonsis said that Apple Computer's 2 Market shopping service on AOL has a \$78 average purchase, which is two times Home Shopping **Network**'s average order, and one-and-a-half times the average paper catalog order. According to Leonsis, 50,000 hours of online shopping time was...

...customer service; and a social dynamic of some kind. For advertising, offer 1) robust interactive information, with a real point of difference; 2) multimedia support -- **text**, graphics, **sound**, and video; 3) a full range of communications options -- e-mail, bulletin boards, and chat; and 4) great customer service (445 Hamilton Avenue, White Plains NY 10601, 914/448-2496). * Dan Burns (former director of Delphi/**Internet**) said that online services are good for providing easy access to "considered" purchases, gifts, and transaction-related products like travel and finance. He said, "Interactive..."

...a critical mass, and marketers have to find and work with good developers who can create the sponsored environments." He opined that establishing an unsupported **site** on the **Internet** would be "like putting a billboard on your lawn, just because there are 100 million cars in the US" (1030 Massachusetts Avenue, Cambridge MA 02138...
...if consumers leave the mass media, advertising as it currently exists will disappear. He noted, "Since Great Britain doesn't have commercial online services, the **Internet** is the center of activity." He reported that 40 percent of online users (including business services) are women. Online Explosion II * Christina Ford (vice president...

...killer interactive applications only when: 1) a McDonald's download doesn't take three hours; 2) the \$30,000 you paid to the Hot Wired **Internet** address actually gets some users to register 3) America Online can be specific about what you get for \$300,000; 4) women dominate; 5) e...

...of interactive advertising conference sessions; and 7) a Time Warner FSN ad doesn't cost a million dollars to reach five homes. Providing advice about **site** creation, she said that "virtual information spaces" require a "diction" that prompts repeat usage, so that consumers want to return to the application. She warned...

18/3,K/5 (Item 3 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)
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02656409 Supplier Number: 45381467 (USE FORMAT 7 FOR FULLTEXT)
THIS WEEK'S LEAD STORY: DYNAMIC ROUTE GUIDANCE DEALT BLOW BY TRIALS OF PHILIPS SOCRATES UNIT
Intelligent Highway, v5, n23, pN/A
March 6, 1995
Language: English Record Type: Fulltext
Document Type: Newsletter; Trade
Word Count: 1302

... every minute, Biding says.
This sizeable discrepancy required the establishment of a new message "filtering process," Biding adds. Travel times for highway links within a **network** allow the unit to calculate routes offering the shortest journey time. The guidance instructions are then provided to the driver by direction arrows on the in-vehicle unit's colour display and through **speech synthesis instructions**.
This data reception problem is confirmed by officials at Volvo, the vehicle manufacturer which supplied vehicles for the trial. The "processing power of the (in...

18/3,K/6 (Item 4 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)
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02289348 Supplier Number: 44426085 (USE FORMAT 7 FOR FULLTEXT)
ND COMTEC BRINGS IN LILLE UNIVERSITY'S PHRASEA MULTIMEDIA ARCHIVING, RETRIEVAL SOFTWARE FOR MAC
Computergram International, n2349, pN/A
Feb 8, 1994
Language: English Record Type: Fulltext
Document Type: Newswire; Trade
Word Count: 648

(USE FORMAT 7 FOR FULLTEXT)

TEXT:

...S, has been appointed distributor and sole UK maintenance provider for Phrasea - a full text indexing multimedia archiving and retrieval software program that manages picture, **text**, video and **sound** files on the same database on the Apple Computer Inc Macintosh. Phrasea automatically indexes and stores information to a free text retrieval database so structured...

...by the University of Lille in Paris, the software was launched in France in October 1993. In the UK, ND Comtec says it will be **pitching** the product at media support industries, newspaper and publishing companies and at local government. Phrasea is available in two versions: Phrasea Agency features only the...

...II is suitable for networks and comprises all the above features. The stand-alone database is GBP1,500 and requires 1.5Mb of RAM, the **server** for networking costs GBP1,725 requiring 3Mb of RAM and an additional 128Kb for each concurrent user; and the communication **server** for remote access is GBP1,100 needing 3Mb of RAM. Phrasea currently runs only on the Apple Macintosh but a Windows version will be available...

18/3,K/7 (Item 1 from file: 148)

DIALOG(R)File 148:Gale Group Trade & Industry DB
(c)2000 The Gale Group. All rts. reserv.

09756784 SUPPLIER NUMBER: 19761691 (USE FORMAT 7 OR 9 FOR FULL TEXT)

Newsroom systems suit up for RTNDA; Windows NT, Web are buzzwords.

(Radio-Television News Directors Association show) (Special Report:

Newsroom Systems)

Dickson, Glen

Broadcasting & Cable, v127, n38, p89(4)

Sep 15, 1997

ISSN: 1068-6827

LANGUAGE: English

RECORD TYPE: Fulltext

WORD COUNT: 3321 LINE COUNT: 00259

... a stand-alone entity at RTNDA. Open systems, not total systems, will be the message in New Orleans for AvidNews, which is designed to handle **text** composition, **audio** and video browsing, and **Web** publishing.

"We understand that lots of customers want a newsroom computer as well as DNG video gear, and some may not want the DNG stuff...

18/3,K/8 (Item 2 from file: 148)

DIALOG(R)File 148:Gale Group Trade & Industry DB
(c)2000 The Gale Group. All rts. reserv.

07218120 SUPPLIER NUMBER: 14984824 (USE FORMAT 7 OR 9 FOR FULL TEXT)

**THE WORLD'S FIRST INTERNET CYBERSTATION TO BROADCAST FROM NETWORLD+INTEROP
94**

PR Newswire, p0407SF007

April 7, 1994

LANGUAGE: ENGLISH

RECORD TYPE: FULLTEXT

WORD COUNT: 648 LINE COUNT: 00057

... forum for addressing the networking interoperability challenges and solutions found in the real world of enterprise computing. The CyberStation will highlight "demonstrated convergence" of voice, **text**, **sound** and image technologies on the **Internet**. The programming includes world news, technical forums and music to be cybercast for the **duration** of the

NetWorld+Interop exhibition. Other highlights from the cybercast include live popular mainstream broadcast programs from National Public Radio (NPR) and news reporting by...
?

22/3,K/1 (Item 1 from file: 275)
DIALOG(R)File 275:Gale Group Computer DB(TM)
(c) 2000 The Gale Group. All rts. reserv.

02048879 SUPPLIER NUMBER: 19244117 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Voice processing: Bell Labs launches Web site for text-to-speech synthesis.
(up to nine different languages) (Company Business and Marketing)
EDGE, on & about AT&T, v12, p23(1)
March 10, 1997
LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 1150 LINE COUNT: 00098

... Romanian.
SPEAKING ON THE WEB Visitors to the Bell Labs Text-to-Speech Synthesis Web site at <http://www.bell-labs.com/project/tts/>, can **sample speech** in up to nine different languages, as well as visit a demonstration area that allows **users** to **synthesize** English, German, and Mandarin Chinese sentences using male, female, and child **intonations** with effects such as raspiness. The **site** offers the experience of high-quality interactive, on-the-fly modifications of **voice samples**.

The Bell Labs TTS system even handles German noun compounds, which are notorious for being long and complex, and which cannot be prestored in a...

22/3,K/2 (Item 1 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)
(c) 2000 The Gale Group. All rts. reserv.

03513997 Supplier Number: 47259439 (USE FORMAT 7 FOR FULLTEXT)
NEW ON THE WEB THIS MONTH...LUCENT TECHNOLOGIES
Internet Business News, pN/A
April 1, 1997
Language: English Record Type: Fulltext
Document Type: Magazine/Journal; Trade
Word Count: 64

(USE FORMAT 7 FOR FULLTEXT)
TEXT:
LUCENT TECHNOLOGIES has introduced its Bell Labs **Text -to-Speech web site** located at <http://www.bell-labs.com/project/tts> designed to allow visitors to product **natural speech** in several languages directly from written text. As well as this, **users** will be able to visit the demonstration section which enables them to **synthesize** English sentences using either male, female or child **intonations**.

22/3,K/3 (Item 2 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)
(c) 2000 The Gale Group. All rts. reserv.

03486605 Supplier Number: 47189608 (USE FORMAT 7 FOR FULLTEXT)
NEW ON THE WEB:LUCENT TECHNOLOGIES
Telecomworldwire, pN/A
March 7, 1997
Language: English Record Type: Fulltext
Document Type: Newsletter; Trade
Word Count: 64

(USE FORMAT 7 FOR FULLTEXT)
TEXT:

LUCENT TECHNOLOGIES has introduced its Bell Labs **Text -to-Speech web site** located at <http://www.bell-labs.com/project/tts> designed to allow visitors to product **natural speech** in several languages directly from written text. As well as this, **users** will be able to visit the demonstration section which enables them to **synthesize** English sentences using either male, female or child **intonations** .

22/3,K/4 (Item 3 from file: 636)
DIALOG(R)File 636:Gale Group Newsletter DB(TM)
(c) 2000 The Gale Group. All rts. reserv.

03485715 Supplier Number: 47187553 (USE FORMAT 7 FOR FULLTEXT)
LUCENT TECHNOLOGIES: Bell Labs launches Web site for Text to Speech synthesis
M2 Presswire, pN/A
March 6, 1997
Language: English Record Type: Fulltext
Document Type: Newswire; Trade
Word Count: 826

... Romanian.

SPEAKING ON THE WEB Visitors to the Bell Labs Text-to-Speech Synthesis Web site at <http://www.bell-labs.com/project/tts/> can **sample speech** in up to nine different languages, as well as visit a demonstration area that allows **users** to **synthesize** English sentences using male, female, and child **intonations** with effects such as raspiness. The **site** offers the experience of high-quality interactive, on-the-fly modifications of **voice samples** .

The Bell Labs TTS system even handles German noun compounds, which are notorious for being long and complex, and which cannot be prestored in a...

22/3,K/5 (Item 1 from file: 148)
DIALOG(R)File 148:Gale Group Trade & Industry DB
(c)2000 The Gale Group. All rts. reserv.

09333788 SUPPLIER NUMBER: 19183995 (USE FORMAT 7 OR 9 FOR FULL TEXT)
Bell Labs Launches Web Site For Text-To-Speech Synthesis.
Business Wire, p3051056
March 5, 1997
LANGUAGE: English RECORD TYPE: Fulltext
WORD COUNT: 1270 LINE COUNT: 00112

... Romanian.

SPEAKING ON THE WEB

Visitors to the Bell Labs Text-to-Speech Synthesis Web site at <http://www.bell-labs.com/project/tts/>, can **sample speech** in up to nine different languages, as well as visit a demonstration area that allows **users** to **synthesize** English, German, and Mandarin Chinese sentences using male, female, and child **intonations** with effects such as raspiness. The **site** offers the experience of high-quality interactive, on-the-fly modifications of **voice samples** .

The Bell Labs TTS system even handles German noun compounds, which are notorious for being long and complex, and which cannot be prestored in a...
?

File 2:INSPEC 1969-2000/Sep W4
 (c) 2000 Institution of Electrical Engineers
 File 6:NTIS 1964-2000/Oct W3
 Comp&distr 2000 NTIS, Intl Cpyrgh All Right
 File 8:EI Compendex(R) 1970-2000/Aug W4
 (c) 2000 Engineering Info. Inc.
 File 14:Mechanical Engineering Abs 1973-2000/Sep
 (c) 2000 Cambridge Sci Abs
 File 65:Inside Conferences 1993-2000/Sep W4
 (c) 2000 BLDSC all rts. reserv.
 File 77:Conference Papers Index 1973-2000/Jul
 (c) 2000 Cambridge Sci Abs
 File 94:JICST-EPlus 1985-2000/May W3
 (c)2000 Japan Science and Tech Corp(JST)
 File 99:Wilson Appl. Sci & Tech Abs 1983-2000/Aug
 (c) 2000 The HW Wilson Co.
 File 108:Aerospace Database 1962-2000/Sep
 (c) 2000 AIAA
 File 144:Pascal 1973-2000/Sep W4
 (c) 2000 INIST/CNRS
 File 233:Internet & Personal Comp. Abs. 1981-2000/Sep
 (c) 2000 Info. Today Inc.
 File 238:Abs. in New Tech & Eng. 1981-2000/Sep
 (c) 2000 Reed-Elsevier (UK) Ltd.
 File 34:SciSearch(R) Cited Ref Sci 1990-2000/Sep W3
 (c) 2000 Inst for Sci Info
 File 434:SciSearch(R) Cited Ref Sci 1974-1989/Dec
 (c) 1998 Inst for Sci Info

Set	Items	Description
S1	3551	((TEXT? ?(2W) (SPEECH OR VOICE)))(5N) SYSTEM? ? OR TTS
S2	2145	TEXT? ? (2N)(TRANSFORM? OR CONVERT? OR CONVERSION? OR SYNT- HES? OR (CHANGE? OR TURN?)(2N)INTO)(5N) (SOUND OR AUDIO? OR V- OICE? OR SPEECH)
S3	17908	(SPEECH OR VOICE) (2N) (SYNTHESES? OR GENERAT?)
S4	20608	S1 OR S2 OR S3
S5	61	S4 AND ((WEB OR NETWORK OR W3 OR INTERNET OR INTRANET)(5N)- (SERVER? OR SITE?) OR WEB() PAGE?)
S6	85	AUDIO(2W)(WAVEFORM? OR WAVE()FORM?)
S7	404	DIGIT?()SEQUENC?
S8	13532	(PROSOD? OR ACCENTUAT? OR INTONATION?)
S9	9333	CONCATENAT?
S10	3241	(SPEECH? OR SOUND? OR VOICE)(2N)(FRAGMENT? OR SAMPL?)
S11	7340	SYLLABLE?
S12	1282	((NATURAL OR HIGH()QUALITY)(3N) (SOUND? OR SPEECH?)) (10- N)(SYNTHESES? OR GENERAT?)
S13	510545	(PITCH? OR DURATION OR APTITUDE OR (ATTACK OR DECAY)(2N) E- NVELOP? OR (SYNTHESES? () INSTRUCT?))
S14	145197	(WAVEFORM? OR WAVE()FORM? ?)
S15	828433	USER? OR CUSTOMER? OR CLIENT? OR SUBSCRIB?
S16	3	S5 AND S6:S14
S17	3495	TEXT? ?(2W) (SPEECH OR VOICE)
S18	2574141	(WEB OR NETWORK OR W3 OR INTERNET OR INTRANET OR SERVER? OR SITE? OR WEB() PAGE?)
S19	690	(S1 OR S17) AND S18
S20	2434882	(SYNTHESES? OR GENERAT?)
S21	138	S19 AND S20

S22 6 S21 AND S5:S6

S23 35161 (ROUT? OR DELIVER? OR SEND OR SENT OR TRANSMIT? OR TRANSMIS? OR PASS? OR REMIT? OR DOWNLOAD)(5N) (AUDIO OR SPEECH OR VOICE OR SOUND)

S24 49 S19 AND S23

S25 29 S24 AND S15

S26 23 RD (unique items)

22/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2000 Institution of Electrical Engineers. All rts. reserv.

6628938 INSPEC Abstract Number: C2000-08-5260S-007

Title: Slovene Interactive Text-to- Speech Evaluation Site- SITES

Author(s): Gros, J.; Mihelic, F.; Pavasic, N.

Author Affiliation: Fac. of Electr. Eng., Ljubljana Univ., Slovenia

Conference Title: Text, Speech and Dialogue. Second International Workshop, TDS'99. Proceedings (Lecture Notes in Artificial Intelligence Vol.1692) p.223-8

Editor(s): Matousek, V.; Mautner, P.; Ocelikova, J.; Sojka, P.

Publisher: Springer-Verlag, Berlin, Germany

Publication Date: 1999 Country of Publication: Germany xi+396 pp.

ISBN: 3 540 66494 7 Material Identity Number: XX-1999-03149

Conference Title: Text, Speech and Dialogue. Second International Workshop, TSD'99. Proceedings

Conference Date: 13-17 Sept. 1999 Conference Location: Plzen, Czech Republic

Language: English

Copyright 2000, IEE

Title: Slovene Interactive Text-to- Speech Evaluation Site- SITES

Abstract: The Slovene Interactive **Text -to-Speech** Evaluation Site (**SITES**) was built according to standards for interactive **speech synthesizer** comparison **sites** as set by COCOSDA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data Consortium). **SITES** aims to give interested listeners a thorough and honest impression of the current **text -to-speech (TTS)** **system** and provides valuable feedback about strong and weak points of the system. The **SITES Web site** enables us to evaluate the S5 Slovene **TTS** system either interactively or off-line by sending the **synthesized speech** file to a given E-mail address. We implemented various standard text selection methods and set up rules for construction as semantically unpredictable sentences for the Slovene language. The evaluation **Web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the user's form input. When the user submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS** system is called with the parameters specified by the user. The **TTS** system **generates** a temporal audio file which is sent back to the user.

...Descriptors: **speech synthesis**

Identifiers: Slovene Interactive **Text -to-Speech** Evaluation Site ; **SITES** ; ...

...**Web site** ; ...

...S5 Slovene **TTS** system...

...**synthesized speech** file

22/3,K/2 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2000 Institution of Electrical Engineers. All rts. reserv.

6482299 INSPEC Abstract Number: B2000-03-6130E-008, C2000-03-5260S-007

Title: SITES: Slovene Interactive Text-to- Speech Evaluation Site

Author(s): Gros, J.; Mihelic, F.; Pavasic, N.

Author Affiliation: Fac. of Electr. Eng., Ljubljana Univ., Slovenia

Conference Title: ISIE '99. Proceedings of the IEEE International

Symposium on Industrial Electronics (Cat. No.99TH8465) Part vol.1 p.
213-16 vol.1

Publisher: IEEE, Piscataway, NJ, USA

Publication Date: 1999 Country of Publication: USA 3 vol. xxiii+1568
pp.

ISBN: 0 7803 5662 4 Material Identity Number: XX-1999-00564

U.S. Copyright Clearance Center Code: 0 7803 5662 4/99/\$10.00

Conference Title: Proceedings of ISIE '99. IEEE International Symposium
on Industrial Electronics

Conference Sponsor: IEEE Ind. Electron. Soc.; Slovenia Ministr. Sci. &
Technol.; Soc. Instrum. & Control Eng. (Japan); Univ. Maribor; Univ.
Ljubljana; IEEE Region 8, Slovenia Sect

Conference Date: 12-16 July 1999 Conference Location: Bled, Slovenia

Language: English

Copyright 2000, IEE

Title: SITES: Slovene Interactive Text-to- Speech Evaluation Site

Abstract: The Slovene Interactive Text -to-Speech Evaluation Site (**SITES**) was built according to standards for interactive **speech synthesiser** comparison **sites** as set by COCOSDA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data Consortium). **SITES** aims to give the interested listeners a thorough and honest impression of the current **text -to-speech (TTS) system** and provides valuable feedback about strong and weak points of the system. The **SITES Web site** enables to evaluate the S5 Slovene **TTS system** either interactively or off-line by sending the **synthesized speech** file to a given e-mail address. We implemented various standard text selection methods and set up rules for construction as semantically unpredictable sentences for the Slovene language. The evaluation **Web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the user's form input. When the user submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS system** is called with the parameters specified by the user. The **TTS system generates** a temporal audio file which is sent back to the user.

...Descriptors: **speech synthesis** ;

Identifiers: Slovene Interactive **Text -to-Speech Evaluation Site** ;
SITES ; ...

...interactive **speech synthesiser** comparison **sites** ; ...

...**text -to-speech system** ; **Web site** ; ...

...S5 Slovene **TTS system**...

...**synthesized speech** file

22/3,K/3 (Item 1 from file: 8)

DIALOG(R)File 8: Ei Compendex(R)

(c) 2000 Engineering Info. Inc. All rts. reserv.

05476840 E.I. No: EIP00025025492

Title: SITES: slovene interactive text-to- speech evaluation site

Author: Gros, Jerneja; Mihelic, France; Pavesic, Nikola

Corporate Source: Univ of Ljubljana, Ljubljana, Slovenia

Conference Title: Proceedings of the 1999 IEEE International Symposium on
Industrial Electronics (ISIE'99)

Conference Location: Bled, Slovenia Conference Date: 19990712-19990716

E.I. Conference No.: 55896

Source: IEEE International Symposium on Industrial Electronics v 1 1999.

p 213-216

Publication Year: 1999

CODEN: 85PTAR

Language: English

Title: SITES: slovene interactive text-to- speech evaluation site

Abstract: The Slovene Interactive Text -to-Speech Evaluation Site (**SITES**) was built according to standards for interactive **speech synthesiser** comparison **sites** as set by COCOSDA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data Consortium). **SITES** aims to give the interested listeners a thorough and honest impression of the current **text -to-speech (TTS) system** and provides valuable feedback about strong and weak points of the system. The **SITES web site** enables to evaluate the S5 Slovene **TTS** system either interactively or off-line by sending the **synthesized speech** file to a given e-mail address. We implemented various standard text selection methods and set up rules for construction as Semantically Unapredictable Sentences for the Slovene language. The evaluation **web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the user's form input. When the user submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS** system is called with the parameters specified by the user. The **TTS** system **generates** a temporal audio file which is sent back to the user. (Author abstract) 15 Refs.

Descriptors: **Speech synthesis** ; Interactive computer systems; Speech intelligibility; World Wide **Web** ; Electronic mail; User interfaces; Speech recognition

Identifiers: Slovene interactive **text to speech** evaluation **site** ; Semantically unapredictable sentence

22/3,K/4 (Item 1 from file: 144)

DIALOG(R)File 144:Pascal

(c) 2000 INIST/CNRS. All rts. reserv.

14317779 PASCAL No.: 99-0525224

Slovene Interactive Text-to- Speech Evaluation site - SITES

TSD '99 : text, speech and dialogue : Plzen, 13-17 September 1999

GROS J; MIHELIC F; PAVESIC N

MATOUSEK Vaclav, ed; MAUTNER Pavel, ed; OCELIKOVA Jana, ed; SOJKA Petr, ed

University of Ljubljana, Faculty of Electrical Engineering, Trzaska 25, 1000 Ljubljana, Slovenia

Text, speech and dialogue. International workshop, 2 (Plzen CZE) 1999-09-13

Journal: Lecture notes in computer science, 1999, 1692 223-228

Language: English

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Slovene Interactive Text-to- Speech Evaluation site - SITES

TSD '99 : text, speech and dialogue : Plzen, 13-17 September 1999

The Slovene Interactive Text -to-Speech Evaluation Site (**SITES**) was built according to standards for interactive **speech synthesiser** comparison **sites** as set by COCOSDA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data Consortium). **SITES** aims to give the interested listeners a thorough and honest impression of the current **text -to-speech (TTS) system** and provides valuable feedback about strong and weak points of the system. The

SITES web site enables to evaluate the S5 Slovene **TTS** system either interactively or off-line by sending the **synthesized speech** file to a given e-mail address. We implemented various standard text selection methods and set up rules for construction as Semantically Unpredictable Sentences for the Slovene language. The evaluation **web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the user's form input. When the user submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS** system is called with the parameters specified by the user. The **TTS** system **generates** a temporal audio file which is sent back to the user.

English Descriptors: **Speech synthesis** ; Interactive system; Slovenian

French Descriptors: **Synthese** parole; **Systeme** conversationnel; Slovene;
Web site ; **Text -to-speech synthesis**

22/3,K/5 (Item 1 from file: 233)
DIALOG(R)File 233:Internet & Personal Comp. Abs.
(c) 2000 Info. Today Inc. All rts. reserv.

00580284 00CX03-002

CT boards' new IP challenge

Grigonis, Richard

Computer Telephony , March 1, 2000 , v8 n3 p120-144, 16 Page(s)

ISSN: 1072-1711

... new IP networks encourage the distribution of telephony resources across the LAN or WAN, making it practical to house media processing (voice compression, DTMF detection/ **generation** , **TTS** , ASR) in a different **server** from **network** interface resources. Notes that along with board components, overall PC systems and CPUs have become more powerful, allowing many CT resource boards to be linked together. Warns that PC-based voice resource vendors might be flanked by data product makers who bolt voice processing into routers and other **network** components. Adds that telecom equipment manufacturers are worried. Describes products from several vendors, along with other ways of accomplishing convergence. Includes nine photos. (KMD)

22/3,K/6 (Item 2 from file: 233)
DIALOG(R)File 233:Internet & Personal Comp. Abs.
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00513254 98IT11-038

CARL's Kid's Catalog moves to the Web

Information Today , November 1, 1998 , v15 n10 p52, 1 Page(s)

ISSN: 8755-6286

Company Name: CARL

URL: <http://www.carl.org>

Product Name: Kid's Catalog **Web**

CARL's Kid's Catalog moves to the Web

Product Name: Kid's Catalog **Web**

Announces the planned release of Kid's Catalog **Web** by the CARL Corporation of Denver, CO (888, 303). Says that the product, now under development, will offer stronger educational and curricular aid with built ...

... be more interactive, enabling users to take notes, compile research

bibliographies, use collaborative learning tools, and publish their projects. Says that full text, online encyclopedias, **Web sites**, and reference tools will be integrated into the product, which will also be compatible with **text -to-speech synthesizers**. Also indicates that it will support Unicode characters in MARC records, making it translatable into any language. (JC)

Descriptors: Children; Reference; Catalog; Online Information;
Information Services; **Speech Synthesis**

Identifiers: Kid's Catalog **Web**; CARL

?

26/3,K/1 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2000 Institution of Electrical Engineers. All rts. reserv.

6628938 INSPEC Abstract Number: C2000-08-5260S-007

Title: Slovene Interactive Text-to- Speech Evaluation Site- SITES

Author(s): Gros, J.; Mihelic, F.; Pavasic, N.

Author Affiliation: Fac. of Electr. Eng., Ljubljana Univ., Slovenia

Conference Title: Text, Speech and Dialogue. Second International Workshop, TDS'99. Proceedings (Lecture Notes in Artificial Intelligence Vol.1692) p.223-8

Editor(s): Matousek, V.; Mautner, P.; Ocelikova, J.; Sojka, P.

Publisher: Springer-Verlag, Berlin, Germany

Publication Date: 1999 Country of Publication: Germany xi+396 pp.

ISBN: 3 540 66494 7 Material Identity Number: XX-1999-03149

Conference Title: Text, Speech and Dialogue. Second International Workshop, TSD'99. Proceedings

Conference Date: 13-17 Sept. 1999 Conference Location: Plzen, Czech Republic

Language: English

Copyright 2000, IEE

Title: Slovene Interactive Text-to- Speech Evaluation Site- SITES

Abstract: The Slovene Interactive **Text -to-Speech** Evaluation **Site (SITES)** was built according to standards for interactive speech synthesizer comparison **sites** as set by COCODA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data Consortium). **SITES** aims to give interested listeners a thorough and honest impression of the current **text -to-speech (TTS) system** and provides valuable feedback about strong and weak points of the system. The **SITES Web site** enables us to evaluate the S5 Slovene **TTS** system either interactively or off-line by sending the synthesized speech file to a given E-mail address. We implemented various standard text selection methods and set up rules for construction as semantically unpredictable sentences for the Slovene language. The evaluation **Web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the **user** 's form input. When the **user** submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS** system is called with the parameters specified by the **user** . The **TTS** system generates a temporal **audio** file which is **sent** back to the **user** .

Identifiers: Slovene Interactive **Text -to-Speech** Evaluation **Site ; SITES ; ...**

...Web site ; ...

...S5 Slovene TTS system

26/3,K/2 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2000 Institution of Electrical Engineers. All rts. reserv.

6497043 INSPEC Abstract Number: B2000-03-6210R-021, C2000-03-6130M-010

Title: CORBA-based multimedia audio chat

Author(s): Cimpu, V.F.; Ionescu, D.; Vieru, V.; Cimpu, M.

Author Affiliation: Sch. of Inf. Technol. & Eng., Ottawa Univ., Ont., Canada

Conference Title: Engineering Solutions for the Next Millennium. 1999 IEEE Canadian Conference on Electrical and Computer Engineering (Cat. No.99TH8411) Part vol.1 p.342-5 vol.1

Editor(s): Meng, M.
Publisher: IEEE, Piscataway, NJ, USA
Publication Date: 1999 Country of Publication: USA 3 vol.
(xxiii+1758) pp.
ISBN: 0 7803 5579 2 Material Identity Number: XX-1999-02278
U.S. Copyright Clearance Center Code: 0 7803 5579 2/99/\$10.00
Conference Title: Engineering Solutions for the Next Millennium. 1999
IEEE Canadian Conference on Electrical and Computer Engineering
Conference Date: 9-12 May 1999 Conference Location: Edmonton, Alta.,
Canada
Language: English
Copyright 2000, IEE

Abstract: This paper presents a chat application that uses CORBA Event and Naming services for communication between **users** and Microsoft **text-to-speech** engines to speak the messages. **Users** can choose the computer voice, which will represent them during the chat, by selecting the gender, speed and pitch of the **text-to-speech** engine. After connecting to a **server**, **users** can create new rooms or browse the existing ones. Before joining a room, a **user** can retrieve other participants' pictures or samples of their real voices. One important feature is the absence of a dedicated chat **server**, which has been replaced by the CORBA Event and Naming services. This allows each host, on which the two CORBA services are running, to be used as a chat **server**. An open message-passing based solution assures synchronization between **users**, as well as the **transmission** of chat messages. A new **Audio** Chat Communication Protocol (ACCP) has been designed for this purpose.
...Identifiers: **text-to-speech** engine...

...chat **server** ;

26/3,K/3 (Item 3 from file: 2)

DIALOG(R) File 2:INSPEC

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6482299 INSPEC Abstract Number: B2000-03-6130E-008, C2000-03-5260S-007

Title: SITES: **Slovene Interactive Text-to-Speech Evaluation Site**
Author(s): Gros, J.; Mihelic, F.; Pavetic, N.
Author Affiliation: Fac. of Electr. Eng., Ljubljana Univ., Slovenia
Conference Title: ISIE '99. Proceedings of the IEEE International Symposium on Industrial Electronics (Cat. No.99TH8465) Part vol.1 p. 213-16 vol.1

Publisher: IEEE, Piscataway, NJ, USA
Publication Date: 1999 **Country of Publication:** USA 3 vol. xxiii+1568 pp.

ISBN: 0 7803 5662 4 **Material Identity Number:** XX-1999-00564
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Conference Title: Proceedings of ISIE '99. IEEE International Symposium on Industrial Electronics
Conference Sponsor: IEEE Ind. Electron. Soc.; Slovenia Minstr. Sci. & Technol.; Soc. Instrum. & Control Eng. (Japan); Univ. Maribor; Univ. Ljubljana; IEEE Region 8, Slovenia Sect
Conference Date: 12-16 July 1999 **Conference Location:** Bled, Slovenia
Language: English
Copyright 2000, IEE

Title: SITES: **Slovene Interactive Text-to-Speech Evaluation Site**
Abstract: The Slovene Interactive **Text-to-Speech Evaluation Site** (**SITES**) was built according to standards for interactive speech synthesiser comparison **sites** as set by COCOSDA (International Committee for the Co-ordination and Standardization of Speech Databases and Assessment Techniques for Speech Input/Output) and the LDC (Linguistic Data

Consortium). **SITES** aims to give the interested listeners a thorough and honest impression of the current **text -to-speech (TTS) system** and provides valuable feedback about strong and weak points of the system. The

SITES Web site enables to evaluate the S5 Slovene **TTS** system either interactively or off-line by sending the synthesized speech file to a given e-mail address. We implemented various standard text selection methods and set up rules for construction as semantically unpredictable sentences for the Slovene language. The evaluation **Web site** has the capability to accept arbitrary input text, and returns a speech file. A CGI script first reads the **user**'s form input. When the **user** submits the form, the script receives the form data as a set of name-value pairs, which is parsed. In the CGI script, the **TTS** system is called with the parameters specified by the **user**. The **TTS** system generates a temporal **audio** file which is **sent** back to the **user**.

Identifiers: Slovene Interactive **Text -to-Speech** Evaluation Site ;

SITES ; ...

...interactive speech synthesiser comparison **sites** ; ...

...**text -to-speech system** ; **Web site** ; ...

...S5 Slovene **TTS** system

26/3,K/4 (Item 4 from file: 2)

DIALOG(R)File 2:INSPEC

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5958775 INSPEC Abstract Number: B9808-6210R-023, C9808-5620W-019

Title: Multimedia digital community: a Web -based multimedia collaboration system

Author(s): Bisdikian, C.; Brady, S.; Doganata, Y.N.; Foulger, D.; Marconcini, F.; Mourad, M.; Operowsky, H.L.; Pacifici, G.; Tantawi, A.N.

Author Affiliation: IBM Thomas J. Watson Res. Center, Yorktown Heights, NY, USA

Conference Title: Fourth IEEE Workshop on High-Performance Communication Systems (HPCS'97) p.57-62

Publisher: HPCS'97 Organization Committee, Chalkidiki, Greece

Publication Date: 1997 Country of Publication: Greece 244 pp.

Material Identity Number: XX97-01491

Conference Title: Proceedings of Fourth Workshop on the Architecture and Implementation of High Performance Communications Subsystems - HPCC'97

Conference Date: 23-25 June 1997 Conference Location: Chalkidiki, Greece

Language: English

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Title: Multimedia digital community: a Web -based multimedia collaboration system

...Abstract: associates on-line. The problem in achieving practical and marketable computer-based multimedia collaboration systems, we believe, has been a lack of standards for non-**voice** multimedia content **delivery** and interaction. However, with the growing usage of the hypertext markup language (HTML) in preparing and linking information on the World-Wide **Web**, a practical base for building standard and broadly available multimedia collaboration solutions is now possible. Realizing that a standards-based, **Web** -enabled conferencing solution could be possible, the idea of a multimedia digital community (MMDC) was conceived with the objective of marrying the desire for on-line interaction and collaboration using **text**, graphics, and **voice** communications, with the **user** -friendliness and pervasiveness of **Web** -based multimedia browser interfaces. MMDC is a **client /server** collaborative solution that has been guided by the need

to develop a system that is open (standards-oriented), platform independent with low barriers of use on the **client** side, and easily migratable onto different platforms and scalable on the **server** side.

...Descriptors: **client -server** systems...

...Internet ;

...Identifiers: **Web** -based multimedia collaboration system...

...World-Wide **Web** ; **Web** -enabled conferencing...

...**client /server** architecture

26/3,K/5 (Item 5 from file: 2)

DIALOG(R)File 2:INSPEC

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03319255 INSPEC Abstract Number: B89020168, C89015337

Title: Comprehensive radiology imaging network: clinical and operational impact

Author(s): Mun, S.K.; Ingeholm, M.L.; Horii, S.; Albers, B.D.

Author Affiliation: Dept. of Radiol., Georgetown Univ. Hospital, Washington, DC, USA

Conference Title: Electronic Imaging '88: International Electronic Imaging Exposition and Conference. Advance Printing of Paper Summaries p.134-8 vol.1

Publisher: Inst. Graphic Commun, Waltham, MA, USA

Publication Date: 1988 Country of Publication: USA 2 vol. xxxviii+1272 pp.

Conference Sponsor: Diagnostic Imaging Magazine; ESD:Electron. Syst. Design Magazine; et al

Conference Date: 3-6 Oct. 1988 Conference Location: Boston, MA, USA

Language: English

Title: Comprehensive radiology imaging network: clinical and operational impact

...Abstract: medical radiologists. In order to test the technical and clinical merit of a functioning IMACS system, Georgetown University has begun the installation of a comprehensive **network** based on AT&T's Comm View System. The Georgetown project is focused on system integration, comprehensive implementation, and diversified **users** ' operation. A comprehensive **network** consists of the following groups: input points: where text and images initially enter the system; **user** workstations: where images are reviewed and reports are generated; communications **network** : consists of image, **text** , and **voice transmission** ; and data storage and database: intermediate data storage and archive devices.

Identifiers: comprehensive radiology imaging **network** ; ...

...comprehensive **network** ; ...

...**user** workstations...

...communications **network** ; **voice transmission** ;

26/3,K/6 (Item 6 from file: 2)

DIALOG(R)File 2:INSPEC

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03286632 INSPEC Abstract Number: B89004497, C89007676

Title: Voice and text messaging-a concept to integrate the services of telephone and data networks

Author(s): Lee, L.-s.; Oun-young, M.

Author Affiliation: Dept. of Electr. Eng., Nat. Taiwan Univ., Taipei, Taiwan

Conference Title: IEEE International Conference on Communications '88: Digital Technology - Spanning the Universe. Conference Record (Cat. No.88CH2538-7) p.408-12 vol.1

Publisher: IEEE, New York, NY, USA

Publication Date: 1988 Country of Publication: USA 3 vol. xxx+1783 pp.

U.S. Copyright Clearance Center Code: CH2538-7/88/0000-0408\$01.00

Conference Sponsor: IEEE

Conference Date: 12-15 June 1988 Conference Location: Philadelphia, PA, USA

Language: English

...Abstract: a voice and text messaging (VTM) system, which can integrate the distinct services of the telephone and data networks very quickly. In Taiwan the telephone **network** has very wide coverage and a large number of **users**, while the data **network** has very limited number of **subscribers**, because they have to possess a terminal. The core of VTM described is a Chinese **text -to- speech system** which can transform any Chinese text processed in the data **network** into Mandarin **voice** for **transmission** over the telephone **network**. The telephone **network users** can key in their instructions such as choice of information, text processing, forward and backward skipping by pressing the touch-tone buttons of the telephone set. The electronic mail and database information services provided by the data **network** therefore become a portion of the voice mail and message services provided by the telephone **network**. A large number of telephone **network users**, even without a terminal, can be served by both networks.

...Identifiers: **network** integration...

...Chinese **text -to-speech system** ;

26/3,K/7 (Item 7 from file: 2)

DIALOG(R) File 2:INSPEC

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03189168 INSPEC Abstract Number: B88052777, C88045977

Title: **An experimental multimedia mail system**

Author(s): Postel, J.B.; Finn, G.G.; Katz, A.R.; Reynolds, J.K.

Author Affiliation: Univ. of Southern California, Marina del Rey, CA, USA

Journal: ACM Transactions on Office Information Systems vol.6, no.1 p.63-81

Publication Date: Jan. 1988 Country of Publication: USA

CODEN: ATOSDO ISSN: 0734-2047

U.S. Copyright Clearance Center Code: 0734-2047/88/0100-0063\$01.50

Language: English

Abstract: With multimedia computer-based mail, a **user** may create messages containing **text**, image, and **voice** data and **send** such messages to other **users** within a computer **network**. The authors describe the development, implementation, and use of one such system. They present an overview of the system, the system model, the presentation model, the multimedia mail program for the **user**'s point of view, and plans for future work.

...Identifiers: computer **network**

26/3,K/8 (Item 8 from file: 2)

DIALOG(R) File 2:INSPEC

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02681182 INSPEC Abstract Number: D86001696

Title: Taking an independent line (telecommunication networks)

Author(s): Horwitt, E.

Journal: Business Computer Systems vol.5, no.2 p.26-32

Publication Date: Feb. 1986 Country of Publication: USA

CODEN: BCOSDI ISSN: 0745-0745

Language: English

...Abstract: deals, price breaks and an expanding menu of services for wide area networks. And the time is right to gear up for Integrated Services Digital **Network** (ISDN), the emerging standard that in a year or so should enable **users** to **send** video images, data, **text** and **voice** over the same digital lines. But it is a difficult time, too, requiring decisions among telecommunications and MIS managers. Not only must they find the...

... combination of communications paths in a wilderness of vendors and options, but they must also decide who assumes responsibility for maintaining, monitoring and managing the **network** -the corporation or the telephone company.

...Identifiers: Integrated Services Digital **Network** ;

26/3,K/9 (Item 9 from file: 2)

DIALOG(R)File 2:INSPEC

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02607476 INSPEC Abstract Number: B86016826, C86013776

Title: The DARPA experimental multimedia mail system

Author(s): Reynolds, J.k.; Postel, J.B.; Katz, A.R.; Finn, G.G.; DeSchon, A.L.

Author Affiliation: Inf. Sci. Inst., Univ. of Southern California, Marina del Rey, CA, USA

Journal: Computer vol.18, no.10 p.82-9

Publication Date: Oct. 1985 Country of Publication: USA

CODEN: CPTRB4 ISSN: 0018-9162

U.S. Copyright Clearance Center Code: 0018-9162/85/1000-0082\$01.00

Language: English

...Abstract: of the Defense Advanced Research Projects Agency are described. This ongoing experiment extends computer mail to include bit map, voice, and other data. With this **system**, **users** can create messages containing **text**, image, and **voice** data and **send** such messages to other **users** in the ARPA **Internet**. Current work focuses on programs and protocols to reach a wider community of **users**.

...Identifiers: ARPA **Internet** ;

26/3,K/10 (Item 10 from file: 2)

DIALOG(R)File 2:INSPEC

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02176402 INSPEC Abstract Number: B84006510, C84005430

Title: Users networks for future offices

Author(s): Necas, J.

Journal: Mechanizace Automatizace Administrativy vol.23, no.9 p. 340-1

Publication Date: 1983 Country of Publication: Czechoslovakia

CODEN: MAUAAU ISSN: 0322-8452

Language: Czech

Title: Users networks for future offices

Abstract: Discusses the development of local area networks (LAN) for future electronic offices. Requirements imposed on networks which enable **transmission** of data, **texts**, pictures and **voice** are discussed and the use of coaxial cables as well as optical fibre cables is considered. Examples of wide-band **user network** adopted in the USA and advanced European Countries are presented and future development trends are discussed. It is concluded that while in the 1980s the...

...Identifiers: wide-band **user network** ;

26/3,K/11 (Item 11 from file: 2)

DIALOG(R)File 2:INSPEC

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02050014 INSPEC Abstract Number: B83031157, C83021193

Title: The structure and operating principles of a 64k-bit/s model network

Author(s): Peter, E.

Journal: Fernmelde-Praxis vol.60, no.3 p.81-94

Publication Date: 10 Feb. 1983 Country of Publication: West Germany

CODEN: FEPXAP ISSN: 0015-0118

Language: German

Title: The structure and operating principles of a 64k-bit/s model network

Abstract: The purpose in the development of this model **network** was to use internationally standardised and compatible techniques not merely for the telephone coverage of large areas but also to handle computer to computer communications, transmission of facsimiles, decentralised printing and high speed data transmission. It will also be used to gather information on the various functions of the **network** and its capabilities. The **subscriber** was provided with a basic 64 kbit/s channel for **transmitting** data, **texts**, or **speech** and a 2.4 kbit/s channel to control the transmission of data and texts. The author concludes that, besides the approach to the operation...

Identifiers: digital **subscriber** loop...

...model **network** ; ...

...**subscriber** ;

26/3,K/12 (Item 12 from file: 2)

DIALOG(R)File 2:INSPEC

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02018860 INSPEC Abstract Number: B83020316

Title: 64-kbit/s switching of text, data and voice using the EDS switching system

Author(s): Hagen, R.

Author Affiliation: Siemens AG, Munich, West Germany

Conference Title: GLOBECOM '82. IEEE Global Telecommunications Conference p.549-52 vol.2

Publisher: IEEE, New York, NY, USA

Publication Date: 1982 Country of Publication: USA 3 vol. xxi+1383 pp.

U.S. Copyright Clearance Center Code: CH1819-2/82-0000-0549\$00.75

Conference Sponsor: IEEE

Conference Date: 29 Nov.-2 Dec. 1982 Conference Location: Miami, FL, USA

Language: English

Title: 64-kbit/s switching of text, data and voice using the EDS switching system

Abstract: The German Post Office (DBP) intends to make a switched digital network with n 64-kbit/s (n<or=4) full-duplex circuits available in 1983. The existing EDS switching nodes in the DBP's Integrated Text and Data Network will be responsible for switching through the bit-sequence-independent connections via which the user will be able to transmit text, data and voice. The signaling for a 64-kbits/s connection is effected, in line with CCITT Recommendation X.21, in a medium-speed out-slot channel. The...

...Identifiers: switched digital network ; ...

...Integrated Text and Data Network ;

26/3,K/13 (Item 1 from file: 6)

DIALOG(R) File 6:NTIS

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2068775 NTIS Accession Number: AD-A340 317/7/XAB

Voice Technology Study Report

(Study rept)

Mogford, R. M. ; Rosiles, A. ; Wagner, D. ; Allendoerfer, K. R.

Federal Aviation Administration Technical Center, Atlantic City, NJ.

Corp. Source Codes: 015213000; 411863

Report No.: DOT/FAA/CT-TN97/2

Dec 97 29p

Languages: English

Journal Announcement: GRAI9814

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NTIS Prices: PC A03/MF A01

This document presents the findings of a voice technology study that evaluated the potential of a speech to text and voice recognition system to support an Airway Facilities maintenance task. Researchers conducted the test at an Airport Surveillance Radar (ASR)-9 site at the William J. Hughes Technical Center. Thirteen Airway Facilities specialists completed the procedure twice, once with the voice technology system and again with a...

... was no more time consuming or difficult to use than a traditional paper manual. The voice recognition rate was 86.6%. Questionnaire responses showed that users found the voice technology system understandable, easy to control, and responsive to voice commands. When asked to compare voice technology to the use of a...

Descriptors: Speech recognition; *Voice communications; *Air traffic control terminal areas; Aircraft maintenance; Performance(Human); Human factors engineering; Feasibility studies; Speech transmission; Man computer interface; Workload; Air traffic controllers; Machine coding; User friendly

26/3,K/14 (Item 2 from file: 6)

DIALOG(R) File 6:NTIS

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1272338 NTIS Accession Number: AD-A173 280/9

ISI (Information Sciences Institute) Experimental Multimedia Mail System
(Research rept)

Postel, J. B. ; Finn, G. G. ; Katz, A. R. ; Reynolds, J. K.

Information Sciences Inst., Marina Del Rey, CA.

Corp. Source Codes: 083386000; 415543

Report No.: ISI/RR-86-173

Sep 86 31p

Languages: English

Journal Announcement: GRAI8703

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NTIS Prices: PC A03/MF A01

With multimedia computer mail, a **user** may create messages containing **text**, image, and **voice** data and **send** such messages to other **users** within a computer **network**. This paper describes the development, implementation, and use of one such system. The following five sections describe the overview of the system, the system model, the presentation model, the multimedia mail program for the **user**'s point of view, and plans for future work. This mail system discusses a computer-based experimental multimedia mail system that allows the **user** to read, create, edit, send, and receive messages containing **text**, images, and **voice**.

Descriptors: Electronic mail; *Computer communications; Message processing; **User** needs; Editing; Facsimile communications; Text processing; Image processing; Voice communications

26/3,K/15 (Item 3 from file: 6)

DIALOG(R)File 6:NTIS

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1228298 NTIS Accession Number: AD-A163 536/6

DARPA (Defense Advanced Research Projects Agency) Experimental Multimedia Mail System

(Research rept)

Reynolds, J. K. ; Postel, J. B. ; Katz, A. R. ; Finn, G. G. ; DeSchon, A. L.

University of Southern California, Marina del Rey. Information Sciences Inst.

Corp. Source Codes: 045598002; 407952

Report No.: ISI/RS-85-164

Dec 85 12p

Languages: English Document Type: Journal article

Journal Announcement: GRAI8610

Pub. in IEEE Computer Magazine, p82-89 Oct 85.

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NTIS Prices: PC A02/MF A01

... describes the development, implementation, and use of an experimental multimedia mail system. About 40 researchers in 10 organizations have contributed to the experiment. With this **system** **users** can create messages containing **text**, image, and **voice** data, and **send** such messages to other **users** in the ARPA **Internet**. Keywords: ARPA **Internet**

; Communication protocols; Computer mail; Electronic mail; and Multimedia.
(Reprints)

26/3,K/16 (Item 4 from file: 6)

DIALOG(R) File 6:NTIS

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1124708 NTIS Accession Number: AD-A143 075/0

Experimental Internetwork Multimedia Mail System

(Research rept)

Katz, A. R.

University of Southern California, Marina del Rey. Information Sciences
Inst.

Corp. Source Codes: 045598002; 407952

Report No.: ISI/RS-84-134

Jun 84 14p

Languages: English

Journal Announcement: GRAI8421

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Springfield, VA, 22161, USA.

NTIS Prices: PC A02/MF A01

This paper describes the implementation and use of an experimental
multimedia mail system, in particular the **user** interface program called
MMM. Using MMM, it is possible for a **user** to create a multimedia message
which may contain various types of **text**, image, and **voice** data and to
then **send** the message to other hosts within the Department of Defense
(DoD) **Internet** Environment. MMM is written in Pascal and runs on a PERQ
personal computer equipped with a large bitmap display, a local hard disk,
and a...

... edited, or others created using a bitmap sketching program (which is
also a part of MMM). Section II of this paper briefly describes the DoD
internet and the family of protocols used in this environment. The
physical data connections between the PERQ running MMM and the various
networks used are also discussed. Section III describes the specific
protocol used. This protocol allows generated types of structured data to
be transferred within the **internet**. Section IV describes the subset of
this protocol implemented in MMM and gives a detailed account of how MMM
works and how one would use...

Descriptors: Message processing; *Data transmission systems; *Computer
communications; Communications networks; Installation; Media; Minicomputers
; **User** needs; Interfaces; Voice communications; Editing

26/3,K/17 (Item 1 from file: 8)

DIALOG(R) File 8:Ei Compendex(R)

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04985517 E.I. No: EIP98044152048

**Title: Telephony based speech technology - from laboratory visions to
customer applications**

Author: Johnston, Denis

Corporate Source: BT Lab, Suffolk, UK

Source: International Journal of Speech Technology v 2 n 2 Dec 1997. p
89-99

Publication Year: 1997

CODEN: ISTEFM ISSN: 1381-2416

Language: English

Title: Telephony based speech technology - from laboratory visions to customer applications

Abstract: This paper describes how research into Automatic Speech Recognition (ASR) and **Text to Speech** Synthesis (**TTS**) is being widely applied within the UK telephone **network**. It compares and contrasts telephony based speech technology with that used in non-telephony based applications and describes some of the special problems associated with integrating these into the existing telephone **network**. In particular, it highlights the main issues concerned with providing flexible, yet robust, multiple channel systems and shows how this has been achieved on a...

Descriptors: Automatic telephone systems; Speech recognition; Speech synthesis; Telecommunication networks; Communication channels (information theory); **Speech transmission**

26/3,K/18 (Item 2 from file: 8)

DIALOG(R)File 8: Ei Compendex(R)

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04785883 E.I. No: EIP97083777126

Title: Experimental Japanese/English interpreting video phone system

Author: Karaorman, Murat; Applebaum, Ted H.; Itoh, Tatsuro; Endo, Mitsuru; Ohno, Yoshio; Hoshimi, Masakatsu; Kamai, Takahiro; Matsui, Kenji; Hata, Kazue; Pearson, Steve; Junqua, Jean-Claude

Corporate Source: Panasonic Technologies, Inc, Santa Barbara, CA, USA

Conference Title: Proceedings of the 1996 International Conference on Spoken Language Processing, ICSLP. Part 3 (of 4)

Conference Location: Philadelphia, PA, USA Conference Date: 19961003-19961006

E.I. Conference No.: 46796

Source: International Conference on Spoken Language Processing, ICSLP, Proceedings v 3 1996. IEEE, Piscataway, NJ, USA, 96TH8206. p 1676-1679

Publication Year: 1996

CODEN: 002642

Language: English

...Abstract: architectural design issues and experiences gained while building and demonstrating an experimental interpreting video phone (IVP) system. The IVP system has been demonstrated in an **internet** home shopping simulation simultaneously before live audiences in Japan and the U.S. An American shop assistant and a Japanese **customer** engaged in task-directed dialogues, using their native languages. In addition to their direct audio/visual contact by ISDN video phone, each participant heard a translation of the remote speaker's utterances in a synthetic voice in real-time. Each **site** used a medium-size vocabulary, a continuous speech recognition **system** and a **text-to-speech** synthesis (**TTS**) **system** for the local language. Recognition results were transmitted over the **internet** to the remote **site**, where the corresponding translated sentence was spoken by **TTS** in the listener's native language. All of the speech and language processing software components of the system were independently developed proprietary technologies of the...

Descriptors: **Speech transmission**; Video telephone equipment; **Speech** synthesis; Linguistics; Wide area networks; Speech recognition

Identifiers: Interpreting video phone (IVP) system; **Internet**

26/3,K/19 (Item 3 from file: 8)

DIALOG(R)File 8: Ei Compendex(R)

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04484828 E.I. No: EIP96083298886

Title: Development of the stand-alone audiotex system

Author: Jeong, Youhyeon; Yi, Sionghun

Corporate Source: Electronics and Telecommunications Research Inst (ETRI), Taejon, S Korea

Conference Title: Proceedings of the 1996 International Conference on Communication Technology Proceedings, ICCT'96. Part 1 (of 2)

Conference Location: Beijing, China Conference Date: 19960505-19960507

E.I. Conference No.: 45212

Source: International Conference on Communication Technology Proceedings, ICCT v 1 1996. IEEE, Piscataway, NJ, USA. p 441-444

Publication Year: 1996

CODEN: 002424

Language: English

Abstract: Audiotex is a general system that combines computers and telephones to **deliver audio** information by adopting **text -to-speech (TTS)** technology. **TTS** is a technology that converts text messages into synthetic speech based on both linguistic analysis of the text and the acoustic knowledge of the production...
...this system, we adopt the pitch synchronous overlap and add (PSOLA) algorithm as the synthesis method. The system is composed of a public switched telephone **network** interface unit, a main control unit, a data interface unit, and **TTS** synthesis unit. It can be applied to a variety of reading services when connected to a host computer and telephone **network**. (Author abstract) 7 Refs.

Descriptors: Telephone systems; Speech synthesis; Information retrieval systems; Information technology; Audio acoustics; Sound reproduction; Computers; Algorithms; Telephone switching equipment; **User** interfaces

Identifiers: Audiotex **system**; Audio information; **Text** to **speech** technology; Speech sounds; Pitch synchronous overlap and add algorithm; Public switched telephone **network**; Data interface unit

26/3,K/20 (Item 4 from file: 8)

DIALOG(R) File 8: Ei Compendex(R)

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04183601 E.I. No: EIP95062743902

Title: Tecnologie vocali interattive sul campo: l'esperienza CSELT

Title: Interactive voice technology at work: the CSELT experience

Author: Billi, R.; Canavesio, F.; Ciaramella, A.; Nebbia, L.

Corporate Source: CSELT

Source: CSELT Technical Reports (Centro Studi e Laboratori Telecomunicazioni) v 23 n 1 Feb 1995. p 75-89

Publication Year: 1995

CODEN: CTRPEJ ISSN: 0393-2648

Language: Italian

...Abstract: paper is a survey of the speech technologies and applications developed at CSELT, some of which are employed in real services on the Italian telephone **network**. With the rise of significant speech recognition and **text -to-speech** applications, the activity of our lab encompasses now a broader set of activities, which range from defining and experimenting new algorithmic approaches to speech product...

...technology research and describes two operative applications, a voice dialing service for large name directories, which is installed in the CSELT PABX, and an automated **network** service for directory assistance, which is now accessible to all the Italian telephone **customers**. (Author abstract)

13 Refs.

Descriptors: **Speech transmission** ; **Voice** /data communication systems;
Speech recognition; Algorithms; Telephone systems; Telecommunication
networks; Automation; Private telephone exchanges

Identifiers: Speech technology; Speech product engineering; Interactive
voice technology; Automated **network** service; Voice dialing service;
Directory assistance

26/3,K/21 (Item 5 from file: 8)

DIALOG(R) File 8:EI Compendex(R)

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03390638 E.I. Monthly No: EI9203031605

Title: Multicast support for group communications.

Author: Ngoh, L. H.

Corporate Source: Univ of Manchester, Manchester, Engl

Source: Computer Networks and ISDN Systems v 22 n 3 Oct 7 1991 p 165-178

Publication Year: 1991

CODEN: CNISE9 ISSN: 0169-7552

Language: English

...Abstract: into existing unicast communication systems to provide
better support for group communications. Multicast services are becoming
more important, as more and more of today's **network** workstation
environments are used to provide group communications for the exchange of
multimedia information left bracket 29 right bracket . These environments
allow **users** to exchange information in the form of 'documents' containing
text , graphics and **voice** ; some **systems** support both store-and-forward
(e.g., mail) and real-time (e.g., conferencing) material. In this paper,
various multicast design issues are addressed and...

...Descriptors: **Voice** /Data Integrated Services; DATA **TRANSMISSION**

26/3,K/22 (Item 1 from file: 99)

DIALOG(R) File 99:Wilson Appl. Sci & Tech Abs

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2104866 H.W. WILSON RECORD NUMBER: BAST00023532

New talk

Bainbridge, Heather;

Wireless Review v. 17 no6 (Mar. 15 2000) p. 18-22

DOCUMENT TYPE: Feature Article ISSN: 1099-9248

ABSTRACT: The marriage of wireless and **Internet** is fueling the
development of voice access to data sources. Voice-recognition and **text**
-to-**speech** services that allow **users** to search a **web site** or check
their e-mail from a wireless phone are being implemented. Some wireless
carriers already offer a service whereby **customers** can dial phone numbers
or navigate their voice mail using voice commands. Internetspeech.com is
beta testing a system that allows **users** to access e-mail and **web sites**
via any telephone. Voice-recognition systems also offer a hands-free
safety factor.

DESCRIPTORS: Integrated **voice** data **transmission** ; ...

...**Internet** telephony;

26/3,K/23 (Item 1 from file: 34)

DIALOG(R) File 34:SciSearch(R) Cited Ref Sci

(c) 2000 Inst for Sci Info. All rts. reserv.

05916066 Genuine Article#: XG279 No. References: 36

Title: Audio/video and synthetic graphics/audio for mixed media

Author(s): Doenges PK (REPRINT) ; Capin TK; Lavagetto F; Ostermann J;
Pandzic IS; Petajan ED

Corporate Source: EVANS & SUTHERLAND COMP CORP, 600 KOMAS DR, POB 58700/SALT
LAKE CITY//UT/84158 (REPRINT); ECOLE POLYTECH FED LAUSANNE, LIG, COMP
GRAPH LAB/CH-1015 LAUSANNE//SWITZERLAND//; UNIV GENOA, DIST, DEPT
TELECOMMUN COMP & SYST SCI/I-16145 GENOA//ITALY//; AT&T BELL LABS, RES
LABS/HOLMDEL//NJ/07733; UNIV GENEVA, CUI, MIRALAB/CH-1211 GENEVA
4//SWITZERLAND//; AT&T BELL LABS, LUCENT TECHNOL/MURRAY HILL//NJ/07974

Journal: SIGNAL PROCESSING-IMAGE COMMUNICATION, 1997, V9, N4 (MAY), P
433-463

ISSN: 0923-5965 Publication date: 19970500

Publisher: ELSEVIER SCIENCE BV, PO BOX 211, 1000 AE AMSTERDAM, NETHERLANDS

Language: English Document Type: ARTICLE (ABSTRACT AVAILABLE)

...Abstract: synthetic, aural and visual (A/V) information. The objective
of this synthetic/natural hybrid coding (SNHC) is to facilitate
content-based manipulation, interoperability, and wider **user** access
in the delivery of animated mixed media, SNHC will support
non-real-time and passive media delivery, as well as more interactive,
real-time...

...streamed A/V objects, and spatial-temporal integration of mixed media
types. Composition, interactivity, and scripting of A/V objects can
thus be supported in **client** terminals, as well as in content
production for **servers**, also more effectively enabling terminals as
servers, Such AIV objects can exhibit high efficiency in transmission
and storage, plus content-based interactivity, spatial-temporal
scalability, and combinations of transient dynamic data and...

...that exploit spatial and temporal coherence over buses and networks.
MPEG-4 responds to trends at home and work to move beyond the paradigm
of **audio** /video as a **passive** experience to more flexible A/V objects
which combine audio/video with synthetic 2D/3D graphics and audio. (C)
1997 Published by Elsevier Science B...

File 350:Derwent WPIX 1963-2000/UD,UM &UP=200046

(c) 2000 Derwent Info Ltd

File 347:JAPIO Oct 1976-2000/May(UPDATED 000915)

(c) 2000 JPO & JAPIO

File 344:Chinese Patents ABS Apr 1985-2000/Aug

(c) 2000 European Patent Office

Set	Items	Description
S1	213	((TEXT? ?(2W) (SPEECH OR VOICE)))(5N) SYSTEM? ? OR TTS
S2	904	TEXT? ? (2N)(TRANSFORM? OR CONVERT? OR CONVERSION? OR SYNT- HES? OR (CHANGE? OR TURN?)(2N)INTO)(5N) (SOUND OR AUDIO? OR V- OICE? OR SPEECH)
S3	1040	S1 OR S2
S4	3	S3 (10N) ((WEB OR NETWORK OR W3 OR INTERNET OR INTRANET)(- 5N)(SERVER? OR SITE?) OR WEB() PAGE?)
S5	220	AUDIO(2W)(WAVEFORM? OR WAVE()FORM?)
S6	1351	(PROSOD? OR ACCENTUAT? OR INTONATION?)
S7	35992	(SPEECH OR VOICE) (2N) (SYNTHE? OR GENERAT?)
S8	2180	CONCATENAT?
S9	1299	(SPEECH? OR SOUND? OR VOICE)(2N)(FRAGMENT? OR SAMPL?)
S11	1263	SYLLABLE?
S13	181328	(PITCH? OR DURATION OR APTITUDE OR (ATTACK OR DECAY)(2N) E- NVELOP? OR (SYNTHE? () INSTRUCT?))
S14	1945	((NATURAL OR HIGH()QUALITY)(3N) (SOUND? OR SPEECH?))
S15	659	TEXT? ?(2W) (SPEECH OR VOICE OR SOUND)
S16	367221	(WEB OR NETWORK OR W3 OR INTERNET OR INTRANET OR SERVER? OR SITE? OR WEB() PAGE?)
S17	129	S15 AND S16
S18	62904	SYNTHE?SIZ?
S19	11	S17 AND S18
S20	7	S17 AND (S5:S6 OR S8:S10 OR S13:S14)
S21	307568	USER? OR CUSTOMER? OR CLIENT? OR SUBSCRIB?
S22	66	S17 AND S21
S23	63	S22 AND (SERVER? OR NETWORK)
S24	37182	(ROUT? OR DELIVER? OR SEND OR SENT OR TRANSMIT? OR TRANS- MIS? OR PASS? OR REMIT?)(5N) (AUDIO OR SPEECH OR VOICE OR - SOUND)
S25	18	(DOWNLOAD)(5N) (AUDIO OR SPEECH OR VOICE OR SOUND)
S26	22	S22 AND (S24 OR S18)
S27	15	S26 NOT S19 NOT S20
S28	310	S15 AND (S18 OR GENERAT?)
S29	85	S28 AND S21
S30	11	S29 AND (S5:S6 OR S8:S10 OR S13:S14)
S31	10	S30 NOT (S19 OR S20)
S32	83469	WAVEFORM? OR WAVE()FORM? ?
S33	36483	S32 AND (S18 OR GENERAT?)
S34	34	S33 AND S15
S35	18	S34 AND (S6 OR S8:S10 OR S13:S14)
S36	17	S35 NOT (S19 OR S20 OR S31)
	?	

?t 4/3,ic,k/1-3

4/3,IC,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

013102007

WPI Acc No: 2000-273878/200024

XRPX Acc No: N00-205313

Communication system between email server and PSTN, to allow subscriber to send and receive messages, using dedicated internet server with text-to- speech conversion

Patent Assignee: KORTX INT SA (KORT-N)

Inventor: AJJAN S; ZANZOURI F

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
FR 2783993	A1	20000331	FR 9811968	A	19980924	200024 B

Priority Applications (No Type Date): FR 9811968 A 19980924

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
FR 2783993	A1	33	H04M-011/00	

International Patent Class (Main): H04M-011/00

Communication system between email server and PSTN, to allow subscriber to send and receive messages, using dedicated internet server with text-to- speech conversion

Abstract (Basic):

... and local equipment (6) connected to the PSTN which can be interrogated by a voice telephone (5). The local equipment can interact with an email **server** (2) via the telephone **network** and store **voice** messages, after **conversion** from **text** format, for subsequent transmission to the telephone subscriber.

... Connection to an email message **server** over the PSTN and **internet** with **conversion** of **text** to **voice** .

4/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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011471102

WPI Acc No: 1997-449009/199741

XRPX Acc No: N97-374168

Accessing and retrieving information from interconnected networks e.g. internet - converting information content of web page from text to speech, signals hyperlink selections of web page into audio manner and allows selection of hyperlinks through use of DTMF signals generated from telephone

Patent Assignee: NETPHONIC COMMUNICATIONS INC (NETP-N)

Inventor: HAHN J S; KWAN R J; OLSEN L E; RHIE K H

Number of Countries: 022 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9732427	A1	19970904	WO 97US3329	A	19970228	199741 B
AU 9719851	A	19970916	AU 9719851	A	19970228	199803
US 5953392	A	19990914	US 96609699	A	19960301	199944

Priority Applications (No Type Date): US 96609699 A 19960301

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9732427 A1 E 57 H04M-002/00

Designated States (National): AU CA JP KR

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC

NL PT SE

AU 9719851 A H04M-001/00 Based on patent WO 9732427

US 5953392 A H04M-001/64

International Patent Class (Main): H04M-001/00; H04M-001/64; H04M-002/00

... converting information content of web page from text to speech, signals hyperlink selections of web page into audio manner and allows selection of hyperlinks through use of DTMF signals generated from telephone

4/3,IC,K/3 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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011373427

WPI Acc No: 1997-351334/199732

XRPX Acc No: N97-291138

Audio access system for resources in wide area network, e.g. Internet - uses audio enabled pages created to link particular text data which can be from WWW and can be retrieved by audio web server for interpreting pages into audio which is displayed at audio interface

Patent Assignee: UNIV RUTGERS STATE NEW JERSEY (RUTF)

Inventor: IMIELINSKI T; VIRMANI A

Number of Countries: 071 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9723973	A1	19970703	WO 96US20409	A	19961220	199732 B
AU 9715664	A	19970717	AU 9715664	A	19961220	199745

Priority Applications (No Type Date): US 959153 A 19951222

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9723973 A1 E 33 H04L-012/16

Designated States (National): AL AM AT AU AZ BB BG BR BY CA CH CN CZ DE

DK EE ES FI GB GE HU IL IS JP KE KG KP KR KZ LK LR LS LT LU LV MD MG MK

MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TM TR TT UA UG US UZ VN

Designated States (Regional): AT BE CH DE DK EA ES FI FR GB GR IE IT KE

LS LU MC MW NL OA PT SD SE SZ UG

AU 9715664 A H04L-012/16 Based on patent WO 9723973

International Patent Class (Main): H04L-012/16

International Patent Class (Additional): H04M-001/64

...Abstract (Basic): system for providing audio access to resources in a wide area network generates an audio enabled page by selectively choosing data from the resources. An audio web server provides text to audio conversion of the audio enabled page. A connection is established to the audio web server from an audio interface. Information is selected and retrieved from the audio enabled page in response to input entered over the connection. The retrieved information...

?

19/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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013247043

WPI Acc No: 2000-418925/200036

XRPX Acc No: N00-313530

**Edit system for telephone message, enables user to correct speech
obtained from speech synthesizer such that corrected speech is provided
as text for transmission over communication system**

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC); IBM CORP (IBMC)

Number of Countries: 002 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 2000148182	A	20000526	JP 99187372	A	19990701	200036 B
CN 1255011	A	20000531	CN 99110989	A	19990702	200045

Priority Applications (No Type Date): US 98185332 A 19981103

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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JP 2000148182	A		32	G10L-015/22	
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CN 1255011	A			H04M-011/00	
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International Patent Class (Main): G10L-015/22; H04M-011/00

International Patent Class (Additional): G06F-017/28; G10L-013/00;

G10L-015/00; H04M-003/42

**Edit system for telephone message, enables user to correct speech
obtained from speech synthesizer such that corrected speech is provided
as text for transmission over communication system**

Abstract (Basic):

... A **server** receives voice input from user through telephone. A
speech-recognition system converts the received voice to a **text** . A
speech-synthesizer coverts the text to a **synthesized** speech to
enable correction by user. The corrected voice is transmitted as text
through a communication system.

... Since corrected speech can be transmitted as **text** , **speech -
synthesizer** is not needed at receiver side to read the corrected
message...

19/3,IC,K/3 (Item 3 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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013071032

WPI Acc No: 2000-242904/200021

XRPX Acc No: N00-183011

**Information processor for e-mail received from portable telephone, has
judging unit to determine skip condition based on output from skip
condition retainer so that mail adapted to skip condition is not read**

Patent Assignee: CANON KK (CANO)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 2000059511	A	20000225	JP 98220808	A	1998080	200021 B

Priority Applications (No Type Date): JP 98220808 A 19980804

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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JP 2000059511 A 8 H04M-003/42
International Patent Class (Main): H04M-003/42
International Patent Class (Additional): G06F-003/16; G06F-013/00;
H04M-011/00

...Abstract (Basic): NOVELTY - The information processor (100) has a mail **server** (104) to manage mail, a mail retainer (102) to hold the currently processing mail and a speech **synthesizer** (103) to convert **text** to **speech**. A skip condition retainer (107) holds skip conditions about the mail as registered by the user. A skip condition judging unit (106) judges the condition...

...skip conditions is not read. DESCRIPTION OF DRAWING(S) - The figure shows block diagram of information processor. (100) Information processor; (102) Mail retainer; (103) Speech **synthesizer**; (106) Judging unit; (107) Skip condition retainer...

19/3,IC,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012889315

WPI Acc No: 2000-061149/200005

XRPX Acc No: N00-047869

Error compensating device for speech data encoding system

Patent Assignee: INT BUSINESS MACHINES CORP (IBM)

Inventor: BANTZ D F; ZAVREL R J

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5987405	A	19991116	US 97881435	A	19970624	200005 B

Priority Applications (No Type Date): US 97881435 A 19970624

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5987405	A	13	G10L-005/00	

International Patent Class (Main): G10L-005/00

International Patent Class (Additional): H04B-001/66

Abstract (Basic):

... signals are converted into digital signals, by A/D converter (10). The digital signals are then converted into text representation by the recognizer (11). The **synthesizer** (14) converts the **text** into original **speech** signal. A compensator (17) synchronizes the original speech signal and facsimile signal by correlation so that the minimum error component is compressed and effective bandwidth...

... For speech data encoding system used in deep space and submarine voice communication, battlefields and in **internet**.

...

...**Synthesizer** (14)

19/3,IC,K/5 (Item 5 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012525605

WPI Acc No: 1999-331711/199928

XRPX Acc No: N99-249346

Call answering method for portable telephone, stationary telephone - involves passing audio to companion based on synthesized modification data

Patent Assignee: HITACHI LTD (HITA)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 11119794	A	19990430	JP 97280054	A	19971014	199928 B

Priority Applications (No Type Date): JP 97280054 A 19971014

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 11119794	A	36	G10L-005/02	

International Patent Class (Main): G10L-005/02

International Patent Class (Additional): G10L-003/00; G10L-003/02; H04M-001/64

... involves passing audio to companion based on synthesized modification data

...Abstract (Basic): NOVELTY - The modification data like sound source set, sound volume parameter, message **text** , **speech** rate are read from the memory based on identified companion. The modification data are **synthesized** and corresponding audio is output to the companion.

DETAILED DESCRIPTION - The key information like name, background sound and telephone number of calling party is extracted...

...USE - For portable telephone, stationary telephone connected to internet .

19/3,IC,K/6 (Item 6 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012371791

WPI Acc No: 1999-177898/199915

XRPX Acc No: N99-131412

Speech synthesis terminal equipment for electronic meeting system - has speech synthesizing unit that converts text information into speech synthesis signal when transmission destination identification information corresponds to identification information

Patent Assignee: SANYO ELECTRIC CO LTD (SAOL)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 11032123	A	19990202	JP 97183372	A	19970709	199915 B

Priority Applications (No Type Date): JP 97183372 A 19970709

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 11032123	A	6	H04M-003/56	

International Patent Class (Main): H04M-003/56

International Patent Class (Additional): G10L-003/00

... has speech synthesizing unit that converts text information into speech synthesis signal when transmission destination identification information corresponds to identification information

...Abstract (Basic): NOVELTY - A speech **synthesizing** unit (36) converts a

text information into a speech synthesis signal when a transmission destination identification information corresponds to the identification information of a receiving...

...equipment. A transmitting data forming unit (33) creates the transmitting data containing the transmission destination identification information. The transmission destination identification information is formed by **synthesizing** the text information and the identification information. A **network** communication unit (34) is used to transmit the created transmitting data to other speech synthesis terminal equipment...

...ADVANTAGE - Enables **synthesizing** the speech of the text information that is sent from the speech synthesis terminal equipment of a transmitting agency. DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of the speech synthesis terminal equipment. (11-14) Speech synthesis terminal equipment; (33) Transmitting data forming unit; (34) **Network** communication unit; (35) Identification information judging unit; (36) Speech **synthesizing** unit...

19/3,IC,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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011916679

WPI Acc No: 1998-333589/199829

XRPX Acc No: N98-260347

Generation method for parametric representation of speech - generating principal set and supplementary set of speech parameters and providing feedback using supplementary set of parameters to modify principal set of parameters

Patent Assignee: MOTOROLA INC (MOTI)

Inventor: CORRIGAN G; KARAALI O; MASSEY N

Number of Countries: 017 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9825260	A2	19980611	WO 97US18815	A	19971015	199829 B
EP 932896	A2	19990804	EP 97946261	A	19971015	199935
			WO 97US18815	A	19971015	

Priority Applications (No Type Date): US 96761627 A 19961205

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9825260 A2 E 28 G10L-000/00

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC NL PT SE

EP 932896 A2 E G10L-005/04 Based on patent WO 9825260

Designated States (Regional): BE DE FR GB

International Patent Class (Main): G10L-000/00; G10L-005/04

...Abstract (Basic): Pref. the modified principal set of speech parameters is output to a waveform **synthesizer** to **synthesize** speech. The coder parameter generating system can be divided into a principal system and a subsystem. The supplementary set of speech parameters consists of energies in each of a predetermined set of frequency bands for speech in a selected time period. The coder parameter generating system can be a neural **network** or a decision tree unit, or alternatively it can use a genetic algorithm...

...USE - For speech synthesis system, e.g. converting **text** to **speech** .

...

...ADVANTAGE - Improves performance of **text -to-speech** system without increasing size of database used to create system

19/3,IC,K/8 (Item 1 from file: 347)

DIALOG(R)File 347:JAPIO

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06562449

EDITING SYSTEM AND METHOD USED FOR TRANSCRIPTION OF TELEPHONE MESSAGE

PUB. NO.: 20-00148182 [JP 2000148182 A]

PUBLISHED: May 26, 2000 (20000526)

INVENTOR(s): MUKUNDO PADOMANABUHAN

MICHAEL PICHENY

DAVID NAHAMUU

SALIM ROOKOSU

APPLICANT(s): INTERNATL BUSINESS MACH CORP <IBM>

APPL. NO.: 11-187372 [JP 99187372]

FILED: July 01, 1999 (19990701)

PRIORITY: 185332 [US 185332], US (United States of America), November 03, 1998 (19981103)

INTL CLASS: G10L-015/22; G06F-017/28; G10L-013/00; G10L-015/00; H04M-003/42

ABSTRACT

PROBLEM TO BE SOLVED: To correct a transcribed **text** with a **voice** by regenerating a **synthesized** speech, making a user correct the **synthesized** voice, and transmitting the corrected voice as a text through a communication system.

SOLUTION: A telephone **server** 26 transfers a text and a diagnosis to a speech **synthesizing server** 34. The speech **synthesizing server** 34 creates a **synthesized** speech and returns this **synthesized** speech to the telephone **server** 26. The telephone **server** 26 regenerates the **synthesized** speech to a user through telephone lines. One purpose of regenerating the **synthesized** speech to the user is to allow the user to correct an unacceptable or inaccurate region. The telephone **server** 26 provides the user with an option of correcting a message. The regeneration of a voice related to a correcting mechanism 36 is achieved in many methods. When the user satisfies the transcription, the telephone **server** 26 transmits the text together with a recorded voice to a message **server** 12.

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19/3,IC,K/9 (Item 2 from file: 347)

DIALOG(R)File 347:JAPIO

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06323595

INFORMATION DISTRIBUTION SYSTEM, INFORMATION TRANSMITTER, INFORMATION RECEIVER AND INFORMATION DISTRIBUTING METHOD

PUB. NO.: 11-265195 [JP 11265195 A]

PUBLISHED: ~~September 28, 1999~~ (19990928)

INVENTOR(s): NAKATSUYAMA TAKASHI

IMAI TSUTOMU

APPLICANT(s): SONY CORP
APPL. NO.: 10-072811 [JP 9872811]
FILED: March 20, 1998 (19980320)
PRIORITY: 5538 [JP 985538], JP (Japan), January 14, 1998 (19980114)
INTL CLASS: G10L-003/00; G06F-003/16; G06F-003/16; G06F-013/00;
G06F-017/28; G10L-005/02

ABSTRACT

... SD). On the side of information receivers 6 and 7, the text information is separated from the intermediate language information and displayed out, voices are **synthesized** while using the intermediate language information, and that synthetic voice information is outputted. Namely, as the intermediate language information, **text** data for **voice synthesization** in voice **synthesizing** processing are analyzed and information made into prescribed data format is transmitted from the **server** side (information transmitters) to the terminal equipment side (information receivers).

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19/3,IC,K/10 (Item 3 from file: 347)
DIALOG(R)File 347:JAPIO
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06308270

VOICE BROWSER SYSTEM

PUB. NO.: 11-249867 [JP 11249867 A]
PUBLISHED: September 17, 1999 (19990917)
INVENTOR(s): NAMIKI IKUO
HAYASHI HIROMICHI
KANAMARU TETSUYA
KIMEDA TSUNEJI
UJIIE MASAMI
APPLICANT(s): NIRPON TELEGR & TELEPH CORP & NTT & NTT ELECTRONICS CORP
APPL. NO.: 10-048180 [JP 9848180]
FILED: February 27, 1998 (19980227)
INTL CLASS: G06F-003/16; G06F-013/00; G06F-013/00

ABSTRACT

... BE SOLVED: To provide a voice browser system which enables even a visually handicapped person to acquire the WWW information.

SOLUTION: This system includes a **server** 100 that has a voice request acquisition means 101 which acquires a request from a client 200 via the input of voices, a voice recognition...

... which transmits a request to the URL that is designated by the client 200 based on the recognition result of the means 102 to an **internet** 70, a voice data generation means 104 which extracts a read-aloud text from the answer given from the **internet** 70 and converts the **text** into the **voice** data to **synthesize** the voices and a voice data transmission means 105 which transmits the voice data generated by the means 104 to the client 200. The system...

... which inputs the requests given from the users in voices, a request issue means 202 which extracts the URL from the result acquired from the **server** 100 and gives a request of an HTML file to the **server** 100 based on the extracted URL and a voice output means 203 which outputs the voice data received from the **server** 100.

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19/3,IC,K/11 (Item 4 from file: 347)
DIALOG(R)File 347:JAPIO
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03944949

ABSENCE GUIDE SYSTEM FOR PRIVATE BRANCH OF EXCHANGE

PUB. NO.: 04-310049 [JP 4310049 A]
PUBLISHED: November 02, 1992 (19921102)
INVENTOR(s): NISHIMORI HISAKIMI
APPLICANT(s): FUJI XEROX CO LTD [359761] (A Japanese Company or
Corporation), JP (Japan)
APPL. NO.: 03-101845 [JP 91101845]
FILED: April 08, 1991 (19910408)
INTL CLASS: [5] H04M-003/42; H04M-003/50; H04Q-003/58
JOURNAL: Section: E, Section No. 1336, Vol. 17, No. 140, Pg. 145,
March 22, 1993 (19930322)

ABSTRACT

... system is provided with work stations 6-1, 6-2, a protocol converter interface processor 2 suited to a communication protocol of a local area **network** 3, a database equipment **server** 4 storing a number of telephone sets 7-1, 7-2 corresponding to the work station connecting to a PBX and an address of the work station with cross reference and a voice **synthesizer** **text** **voice** conversion section 5 converting text information into a voice signal.
?

20/3,IC,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012914677

WPI Acc No: 2000-086513/200007

Related WPI Acc No: 1999-046193

XRPX Acc No: N00-067916

Remote monitoring method of interaction between call center attendant and caller in telecommunication system

Patent Assignee: METRO ONE TELECOM INC (METR-N)

Inventor: COX P M; GIRSCH J E; HUEY C A; KEPLER M A; LEE A S; POWELL A P

Number of Countries: 085 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9959316	A1	19991118	WO 99US10268	A	19990511	200007 B
AU 9939803	A	19991129	AU 9939803	A	19990511	200018

Priority Applications (No Type Date): US 9875780 A 19980511

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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WO 9959316	A1	E	48	H04M-003/00	
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Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN
CU CZ DE DK EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ
LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK
SL TJ TM TR TT UA UG UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR
IE IT KE LS LU MC MW NL OA PT SD SE SL SZ UG ZW

AU 9939803	A			H04M-003/00	Based on patent WO 9959316
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International Patent Class (Main): H04M-003/00

International Patent Class (Additional): H04L-012/66

Abstract (Basic):

... identification, destination party identification, geographical origination and destination of the call, date and time of the call, service provider, call center, call center attendant and **duration** of the call. An INDEPENDENT CLAIM is also included for remote monitoring apparatus between call center attendant and caller in telecommunication system...

...in the call monitor. The interface with which the reviewer is connected, allows reviewer to access call recordings stored in the call monitor via a **web** browser or other interfaces, to enable speech recognition, speech-to-text conversion, **text** -to-**speech** conversion and to obtain information displayed on the call center attendant's terminal during the call etc...

20/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012753664

WPI Acc No: 1999-559781/199947

XRPX Acc No: N99-413378

Speech signal distribution system for computer network

Patent Assignee: LERNOUT & HAUSPIE SPEECHPRODUCTS (LERN-N)

Inventor: TEL M P

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5943648	A	19990824	US 96638061	A	19960425	199947 B

Priority Applications (No Type Date): US 96638061 A 19960425

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5943648	A		11 G10L-005/02	

International Patent Class (Main): G10L-005/02

Speech signal distribution system for computer network

Abstract (Basic):

... **Text -speech** parameter converter converts text containing sentences into a data stream with speech signal parameters representing spoken text and lacking phrase sentence level **prosodic** content. A supplemental parameter generator (128) inserts additional data representing linguistic boundaries which represent parameters associated with predefined boundaries into the data stream.

... For computer **network** including **Internet** for transmitting voice messages in encoded form and for generating animated pictures of a person speaking simultaneously with corresponding audio signal...

...Title Terms: **NETWORK**

20/3,IC,K/3 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012481050

WPI Acc No: 1999-287158/199924

XRPX Acc No: N99-214450

Speaker access control method using text independent speech recognition e.g. for banking services

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)

Inventor: KANEVSKY D; MAES S H

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5897616	A	19990427	US 97871784	A	19970611	199924 B

Priority Applications (No Type Date): US 97871784 A 19970611

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5897616	A		15 G10L-009/08	

US 5897616 A 15 G10L-009/08

International Patent Class (Main): G10L-009/08

Speaker access control method using text independent speech recognition e.g. for banking services

Abstract (Basic):

... A **voice sample** is taken from the utterances and processed against an acoustic model. A score corresponding to accuracy of decoded answer and closeness of match between **voice sample** and acoustic model. The score is compared to predefined threshold value and when above it, speaker access to the **server** is permitted. An INDEPENDENT CLAIM is also included for speaker access control apparatus...

20/3,IC,K/4 (Item 4 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012297223

WPI Acc No: 1999-103329/199909

Intonation generation method of a text-to- speech conversion system using intonation pattern normalization and neural network learning - NoAbstract

Patent Assignee: KOREA ELECTRONICS & TELECOM RES (KOEL-N)

Inventor: HAN M S; KIM S H; LEE J C; LEE Y J

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
KR 97050108	A	19970729	KR 9555841	A	19951223	199909 B

Priority Applications (No Type Date): KR 9555841 A 19951223

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
KR 97050108	A		G10L-005/00	

International Patent Class (Main): G10L-005/00

Intonation generation method of a text-to- speech conversion system using intonation pattern normalization and neural network learning...

Title Terms: **INTONATION** ;

20/3,IC,K/5 (Item 5 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012273659

WPI Acc No: 1999-079765/199907

XRPX Acc No: N99-057432

Text to speech profile interchange for text message chatting - uses the interchanging of the text to speech profile with inclusion of control code in the text message

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
RD 416110	A	19981210	RD 98416110	A	19981120	199907 B

Priority Applications (No Type Date): RD 98416110 A 19981120

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
RD 416110	A	1	G06F-000/00	

International Patent Class (Main): G06F-000/00

Text to speech profile interchange for text message chatting...

...uses the interchanging of the text to speech profile with inclusion of control code in the text message

...Abstract (Basic): Operation of the system commences once a **network** conversation connection between another person is commenced. The system swaps the **Text** to **speech** (TTS) profile to represent the character of the person who is speaking on the opposite side. Characteristics available include male or female tone, frequency and **pitch** of speaker, and volume of the **intonation** .

...

...ADVANTAGE - Reduces the system and **network** traffic and offers a text orientated human readable file

20/3,IC,K/6 (Item 6 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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011855597

WPI Acc No: 1998-272507/199824

XRPX Acc No: N98-213896

**Generation of segment durations in text-to- speech system - mapping
sequence of phones to sequence of articulatory features, using prominence
and boundary information as well as predetermined set of rules for type,
phonetic context and syntactic and prosodic context**

Patent Assignee: MOTOROLA INC (MOTI)

Inventor: CORRIGAN G; KARAALI O; MASSEY N

Number of Countries: 018 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9819297	A1	19980507	WO 97US18761	A	19971015	199824 B
EP 876660	A1	19981111	EP 97946842	A	19971015	199849
			WO 97US18761	A	19971015	
US 5950162	A	19990907	US 96739975	A	19961030	199943

Priority Applications (No Type Date): US 96739975 A 19961030

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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WO 9819297	A1	E	24	G10L-003/02	
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Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC
NL PT SE

EP 876660	A1	E	G10L-003/02	Based on patent WO 9819297
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Designated States (Regional): BE DE FR GB

US 5950162	A	G10L-005/06
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International Patent Class (Main): G10L-003/02; G10L-005/06

International Patent Class (Additional): G10L-009/00

Generation of segment durations in text-to- speech system...

**...phones to sequence of articulatory features, using prominence and
boundary information as well as predetermined set of rules for type,
phonetic context and syntactic and prosodic context**

...Abstract (Basic): The method for generating segment durations in a **text
-to-speech** system comprises generating an information vector for each
segment description. The information vector includes a description of a
sequence of segments surrounding described segment and...

**...The information vector is supplied as an input to a pre-trained neural
network . A description is generated representing the duration
associated with the described segment...**

...ADVANTAGE - Avoids effects when **network** depends on chance correlations
in training data and provides efficient segment durations...

...Title Terms: DURATION ;

20/3,IC,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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010491794

WPI Acc No: 1995-393195/199550

XRPX Acc No: N95-286661

Text conversion method for generating audible signals using neural network - training neural network to associate text of recorded spoken messages with speech of spoken messages by converting recorded spoken messages into series of audio frames of fixed duration

Patent Assignee: MOTOROLA INC (MOTI)

Inventor: CORRIGAN G E; GERSON I A; KARAALI O

Number of Countries: 022 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9530193	A1	19951109	WO 95US3492	A	19950321	199550 B
FI 9505608	A	19951122	WO 95US3492	A	19950321	199607
			FI 955608	A	19951122	
AU 9521040	A	19951129	AU 9521040	A	19950321	199609
EP 710378	A1	19960508	EP 95913782	A	19950321	199623
			WO 95US3492	A	19950321	
JP 8512150	W	19961217	JP 95528216	A	19950321	199710
			WO 95US3492	A	19950321	
AU 675389	B	19970130	AU 9521040	A	19950321	199713
US 5668926	A	19970916	US 94234330	A	19940428	199743
			US 96622237	A	19960322	
CN 1128072	A	19960731	CN 95190349	A	19950321	199750
CA 2161540	C	20000613	CA 2161540	A	19950321	200042
			WO 95US3492	A	19950321	

Priority Applications (No Type Date): US 94234330 A 19940428; US 96622237 A 19960322

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 9530193	A1	E	40	G06F-015/18	
					Designated States (National): AU CA CN FI JP
					Designated States (Regional): AT BE CH DE DK ES FR GB GR IE IT LU MC NL PT SE
FI 9505608	A			G10L-000/00	
AU 9521040	A			G06F-015/18	Based on patent WO 9530193
EP 710378	A1	E	40	G06F-015/18	Based on patent WO 9530193
					Designated States (Regional): DE FR GB SE
JP 8512150	W		40	G10L-003/00	Based on patent WO 9530193
AU 675389	B			G06F-015/18	Previous Publ. patent AU 9521040
					Based on patent WO 9530193
US 5668926	A		19	G10L-005/06	Cont of application US 94234330
CN 1128072	A			G06F-015/18	
CA 2161540	C	E		G10L-005/04	Based on patent WO 9530193
					International Patent Class (Main): G06F-015/18; G10L-000/00; G10L-003/00; G10L-005/04; G10L-005/06

Text conversion method for generating audible signals using neural network - ...

...training neural network to associate text of recorded spoken messages with speech of spoken messages by converting recorded spoken messages into series of audio frames of fixed duration

...Abstract (Basic): of converting text into audible signals involves using recorded audio messages (204) which are converted into a series of audio frames (205) having a fixed duration (213). Each audio frame is assigned a phonetic representation (203) and a target acoustic representation. The phonetic representation is (203) is a binary word that represents the phone and articulation characteristics of the audio frame. The target representation is a vector of audio information such as pitch and energy...

...After training, the neural **network** is used in conversion of **text** into **speech**. Text that is to be converted is translated into a series of phonetic frames of the same form as phonetic representations (203) and having a fixed **duration** (213). The neural **network** then produces acoustic representations in response to context descriptions (207) that include some of the phonetic frames. The acoustic representations are then converted into speech...

...Abstract (Equivalent): A method for training and utilizing a neural **network** that is used to convert text streams into audible signals, the method comprising the steps of...

...wherein training a neural **network** utilizes the steps of...

...1b) dividing the recorded audio messages into a series of audio frames, wherein each audio frame has a fixed **duration** ;

...

...1f) training a feed-forward neural **network** with a recurrent input structure to associate an acoustic representation of the plurality of acoustic representations with the context description of the each audio frame...

...a phonetic frame of the series of phonetic frames includes one of the plurality of phonetic representations, and wherein a phonetic frame has the fixed **duration** ;

...

...1i) converting, by the neural **network** , the phonetic frame into one of the plurality of acoustic representations, based on the one of the plurality of context descriptions; and

...Title Terms: **NETWORK** ;

?

?t /3,ic,k/1-15

27/3,IC,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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013293146

WPI Acc No: 2000-465081/200040

XRPX Acc No: N00-347160

Communication method for use in IP-based telephone communication, involves converting voice data and selectively generated voice text to packetized signal which is then transmitted over packet switched network

Patent Assignee: ERICSSON INC (TELF)

Inventor: HIRI F

Number of Countries: 089 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200033552	A1	20000608	WO 99US28215	A	19991129	200040 B
AU 200017472	A	20000619	AU 200017472	A	19991129	200044

Priority Applications (No Type Date): US 98200879 A 19981130

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200033552 A1 E 22 H04M-007/00

Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN
CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP
KR KZ LC LK LR LS LT LU LV MA MD MG MK MN MW MX NO NZ PL PT RO RU SD SE
SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR
IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW

AU 200017472 A H04M-007/00 Based on patent WO 200033552

International Patent Class (Main): H04M-007/00

International Patent Class (Additional): H04L-012/64

... use in IP-based telephone communication, involves converting voice data and selectively generated voice text to packetized signal which is then transmitted over packet switched network

Abstract (Basic):

... data is then processed and applied with a work list. One or more speech patterns within the voice data is recognized to selectively generate voice **text** . The **voice** data and voice text are converted to packetized signal which is then transmitted over a packet switched **network** .

... In IP based telephone communication computer networks such as **internet** .

...Due to connection between two or more PCs over the **internet** , audio and video data generated in one PC is packetized and transported over the **internet** for display on the other PC, so **users** may view each other while simultaneously speaking to each other. Allows caller to view received video data while concurrently **transmitting** video and **speech** generated data

...Title Terms: **NETWORK**

27/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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013192316

WPI Acc No: 2000-364189/200031

XRPX Acc No: N00-272527

Information delivery system for Internet based subscriber network, updates information in playback device according to subscriber preferences, when device gets disconnected from subscriber PC

Patent Assignee: LEXTRON SYSTEMS INC (LEXT-N)

Inventor: KIKINIS D

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 6055566	A	20000425	US 985562	A	19980112	200031 B

Priority Applications (No Type Date): US 985562 A 19980112

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 6055566	A	8	G06F-015/16	

International Patent Class (Main): G06F-015/16

Information delivery system for Internet based subscriber network, updates information in playback device according to subscriber preferences, when device gets disconnected from subscriber PC

Abstract (Basic):

... A **subscriber** PC (123) downloads text documents from **Internet** connected host **server** (120). When playback device (110) is connected to PC, text documents are stored. The device renders the **text** documents, as **speech** on-demand, when disconnected from PC. A radio broadcast unit and a receiver updates information in the device, according to **subscriber** preferences, when the device is disconnected from the PC.

... A host **server** (120) compiles information, stores **subscriber** preferences and sorts information. The **server** adjusts stored **subscriber** preferences in accordance with **subscriber** use patterns and delivers information, as text documents through **Internet** (100). The host **server** codes text documents **delivered** to **subscriber** for controlling **audio** characteristics including inflection. An INDEPENDENT CLAIM is also included for multimedia information output procedure...

...For providing various multimedia data to PC **subscribers** through **Internet** .

...Facilitates connection of localized media sources, since a digital **network** can be replicated along with host **server** and can be distributed to different usage areas...

...The figure shows over view diagram of **Internet** -based media delivery system...

...**Internet** (100...

...Host **server** (120...

...**Subscriber** PC (123

...Title Terms: **SUBSCRIBER ; NETWORK ;**

27/3,IC,K/3 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012934138

WPI Acc No: 2000-105985/200009

XRPX Acc No: N00-081397

Electronic message delivering system e.g. for e-mail, voice mail for digital mobile phones

Patent Assignee: LOGICA INC (LOGI-N)

Inventor: FERNANDEZ D E; HAYDEN B; HUDSON M; PETRIE D G

Number of Countries: 019 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9965256	A2	19991216	WO 99US13183	A	19990610	200009 B

Priority Applications (No Type Date): US 9888781 A 19980610

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
WO 9965256	A2	E 34	H04Q-007/00	

Designated States (National): JP

Designated States (Regional): AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

International Patent Class (Main): H04Q-007/00

Abstract (Basic):

... The **user** selected data is retrieved from the e-mail address of the **user** via **internet** through dial-up connection, LAN. The message is filtered by the **user** specified configurations and summarized with a message identifier. The message is then delivered to the **user** by message **network** such as public switched telephone **network** (PSTN).

... The system consists of a digitized interactive voice response (IVR) capable of receiving message identifier and **user** instructions via data delivery interface protocols like SMTP, TAP etc. **Text** to **speech** system is provided for converting message **text** to **speech** for playing back message on **user** request. Reply e-mails with address derived from the identified e-mail can be **sent** through the **voice** mail notification **server**. The retrieval system repeatedly polls the **user** e-mail address for new messages where the polling depends on the e-mail activity. An INDEPENDENT CLAIM is also included for electronic message delivering...

...Used for **delivering** messages such as e-mail, **voice** mail to digital mobile phones...

...Achieves immediate notification of e-mail arrivals due to the repeated polling of e-mail address. Offers option to get data in **text** or in **speech** format due to usage of IVR. Selection of message is made possible by using filtering...

27/3,IC,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012865024

WPI Acc No: 2000-036857/200003

XRPX Acc No: N00-027633

Transmitting information over mobile telephone network by general broadcasting - provides information to telephone users over restricted geographic region, including numbers which can be dialled for further information

Patent Assignee: TELIA AB (TELI-N)

Inventor: EMILSSON S

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
SE 9801267	A	19991010	SE 981267	A	19980409	200003 B

Priority Applications (No Type Date): SE 981267 A 19980409

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
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SE 9801267	A	10	H04M-011/08	
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International Patent Class (Main): H04M-011/08

International Patent Class (Additional): H04H-001/00; H04H-009/00;
H04Q-007/22

Transmitting information over mobile telephone network by general broadcasting...

...provides information to telephone users over restricted geographic region, including numbers which can be dialled for further information

...Abstract (Basic): NOVELTY - The broadcast provides information which needs to be delivered to a large number of mobile telephone **users**, is carried out over a restricted geographic region, and contains information on telephone numbers that can be dialled to receive further information. **IMAGING and COMMUNICATION - PREFERRED FEATURES** : The information is short text-based and is sent over a GSM **network** using a short message service cell broadcast (SMSCB) system...

...ADVANTAGE - **Text** or **voice** information can be **sent** directly by e.g. a seller, club or organisation to a large number of mobile telephone **users** .

...Title Terms: NETWORK ;

27/3,IC,K/5 (Item 5 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012754110

WPI Acc No: 1999-560227/199947

XRPX Acc No: N99-413818

Conversant-type voice recognition and command process for computer communication from remote location

Patent Assignee: LUCENT TECHNOLOGIES INC (LUCE)

Inventor: YAKER R

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5950167	A	19990907	US 9813665	A	19980126	199947 B

Priority Applications (No Type Date): US 9813665 A 19980126

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
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US 5950167	A	15	G10L-009/06	
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International Patent Class (Main): G10L-009/06

Abstract (Basic):

... A **user** -entered tone and **voice** signals **transmitted** from a telephone (21) to a controller (15) are converted as application-specific commands which are executed by a processor. The **user** is prompted with voiced queries in a VCS (16) to issue sequenced commands. The **user** interrupts an ongoing application program **routine** with **voice** commands to invoke new application program functions.

... A voice command system (VCS) (16) consists of a voice recognition unit (VRU) (17), a voice to text and **text** to **voice** converter (18). The controller (15) connects the VCS and a personal computer (1) to a telephone **network** (20). The voice to text converter consists of a software for converting voice commands and tone signals to application program-specific commands. The signals include...

...printer, copier, facsimile, or an e-mail address. The processor executes the commands under the control of the controller to perform application program functions. The **user** interrupts the ongoing application program such as word processor (12) with voiced commands to invoke the new application program such as spread sheet (13), e...

...The ability of a **user** to direct application program files on personal computer to a destination, by remotely- issued tone or voice commands greatly enhances the utility of personal computers...

...The figure shows the block diagram of the controller, VCS and personal computer connected to the telephone **network** .

...

...Telephone **network** (20)

27/3,IC,K/6 (Item 6 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012712162

WPI Acc No: 1999-518275/199943

XRPX Acc No: N99-385451

Self-contained intelligent radio for receiving broadcasts from both local radio stations and world wide web WWW

Patent Assignee: QURESHEY S (QURE-I); QURESHEY W (QURE-I)

Inventor: QURESHEY S; QURESHEY W

Number of Countries: 083 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9938266	A1	19990729	WO 99US1001	A	19990119	199943 B
AU 9923240	A	19990809	AU 9923240	A	19990119	200001

Priority Applications (No Type Date): US 9896703 A 19980612; US 9872127 A 19980122

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
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WO 9938266	A1	E 32	H04B-001/06	
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Designated States (National): AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SZ UG ZW

AU 9923240 A H04B-001/06 Based on patent WO 9938266

International Patent Class (Main): H04B-001/06

Self-contained intelligent radio for receiving broadcasts from both local radio stations and world wide web WWW

Abstract (Basic):

... A stored software program is configured to connect a modem (206) to an **Internet** service provider and receive digitized audio

broadcasts from the **Internet** service provider. The program is further configured to provide a select broadcast display that allows a **user** to selectably connect a program broadcast to the input of an audio amplifier (222) from the AM or FM radio station or the WWW.

... A display device (11) provides information to the **user** . A tuning control (114) is operated to receive radio frequency RF signals from the radio broadcast stations. The stereo speakers (106,108) are operably connected to the **audio** amplifier. The modem **transmits** and receives digital data over a communications **network** . A data storage device (210) stores the software program...

...Can be used for **Internet** telephony, voicemail, **text** -to-voice mail, voice-to-text electronic mail and voice activated commands...

...Allows **user** to receive **Web** radio broadcasts in a manner similar to the ease and low cost with which the **user** receives regular radio broadcasts. Relieves **user** of complicated tasks associated with installing and configuring computer software since **user** interface that is less like computer program and more like conventional radio is provided, thereby making radio easy to use. **User** can tune into **Web** , AM or FM broadcast with ease through tuning control. Has lower cost, smaller size, lower power consumption, less upkeep and maintenance and more convenience compared with full-fledged computer. Provides hardware and software necessary to receive digitized radio from **Web** without need for personal computer or other expensive equipment...

...Title Terms: **WEB**

27/3,IC,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012337095

WPI Acc No: 1999-143202/199912

XRPX Acc No: N99-104021

Method for delivering electronic mail message from remote source to subscriber station - receives and stores email message, sends signal to subscriber station indicating message is waiting retrieval, sends request to read message, retrieves waiting message, converts it into speech message and sends this to subscriber station

Patent Assignee: ERICSSON INC (TELF)

Inventor: NELSON M P

Number of Countries: 081 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9905626	A1	19990204	WO 98US14974	A	19980720	199912 B
AU 9886591	A	19990216	AU 9886591	A	19980720	199926
US 6061718	A	20000509	US 97899772	A	19970723	200030

Priority Applications (No Type Date): US 97899772 A 19970723

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
WO 9905626	A1	17	G06F-017/60	

Designated States (National): AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DK EE ES FI GB GE GH GM HU ID IL IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SZ UG ZW

AU 9886591 A G06F-017/60 Based on patent WO 9905626

US 6061718 A G06F-013/38

International Patent Class (Main): G06F-013/38; G06F-017/60

International Patent Class (Additional): G06F-015/17

Method for delivering electronic mail message from remote source to subscriber station...

...receives and stores email message, sends signal to subscriber station indicating message is waiting retrieval, sends request to read message, retrieves waiting message, converts it into speech message and sends this to subscriber station

...Abstract (Basic): NOVELTY - The electronic email delivery system (44 to 50) delivers email messages to and from a **subscriber** station (30) in a wireless system. The system converts the messages sent to the **subscriber** station from **text** to **speech**. The **delivery** system converts the email messages sent by the **subscriber** station from **speech** to text for **delivery** to a remote destination. DETAILED DESCRIPTION - **Subscriber** station is a mobile station and the message waiting signal is sent on an analog or digital control channel in the system...

...USE - For delivering electronic mail messages in wired or wireless communications system, messages are of unrestricted length and sent to fixed or mobile **subscriber** who can learn contents of messages without being distracted from performing other activities...

...ADVANTAGE - System does not restrict the length of the email message to a mobile **subscriber** and allows the **subscriber** to learn the contents of the message without being distracted from performing other activities. DESCRIPTION OF DRAWING(S) - The drawing shows a block diagram of an email delivery system. (44) email **server**; (50) base station; (30) **subscriber** station...

...Title Terms: **SUBSCRIBER** ;

27/3,IC,K/8 (Item 8 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012275506

WPI Acc No: 1999-081612/199907

XRPX Acc No: N99-058697

Information transmission method for telecommunications networks - has subscriber requesting information and response text-to- speech coded with telephone access point signal conversion.

Patent Assignee: TELECOM PTT FORSCHUNG & ENTWICKLUNG (TELE-N); SWISSCOM AG (SWIS-N)

Inventor: VAN KOMMER R

Number of Countries: 079 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9859486	A1	19981230	WO 97CH246	A	19970620	199907 B
AU 9730864	A	19990104	AU 9730864	A	19970620	199921
			WO 97CH246	A	19970620	
EP 993730	A1	20000419	EP 97925810	A	19970620	200024
			WO 97CH246	A	19970620	

Priority Applications (No Type Date): WO 97CH246 A 19970620

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9859486 A1 F 35 H04M-003/50

Designated States (National): AL AM AT AU AZ BA BB BG BR BY CA CH CN CU

CZ DE DK EE ES FI GB GE GH HU IL IS JP KE KG KP KR KZ LC LK LR LS LT LU
LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT UA
UG US UZ VN YU ZW

Designated States (Regional): AT BE CH DE DK EA ES FI FR GB GH GR IE IT
KE LS LU MC MW NL OA PT SD SE SZ UG

EP 993730 A1 F H04M-003/50 Based on patent WO 9859486

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LI LU
MC NL PT SE

AU 9730864 A H04M-003/50 Based on patent WO 9859486

International Patent Class (Main): H04M-003/50

... has subscriber requesting information and response text-to- speech
coded with telephone access point signal conversion.

...Abstract (Basic): The information transmission method has a **subscriber**
making a local telephone call to a telephone information service (1),
for instance a weather forecast...

...The information is coded in semantic form using **Text** to **Speech**
conversion (TTS) and **transmitted** over the transmission **network**
(10). Prior to the **subscriber** telephone (30) there is a convertor (2)
which converts the text format to digital words for normal telephone
reception...

...ADVANTAGE - The transmission of the information using semantic code
reduces transmission bandwidth and thus loading the **network** less than
previous systems...

...Title Terms: **NETWORK ; SUBSCRIBER ;**

27/3,IC,K/9 (Item 9 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012051070

WPI Acc No: 1998-467980/199840

XRPX Acc No: N98-364682

**Announcement provision method in communication network - using service
control point in evaluation of supportability of announcements by unit
which converts received text into message for caller**

Patent Assignee: SIEMENS AG (SIEI)

Inventor: NIMPHIUS K

Number of Countries: 022 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9837716	A2	19980827	WO 98DE377	A	19980211	199840 B
EP 962106	A2	19991208	EP 98910604	A	19980211	200002
			WO 98DE377	A	19980211	
CN 1248377	A	20000322	CN 98802749	A	19980211	200032
BR 9807258	A	20000523	BR 987258	A	19980211	200035
			WO 98DE377	A	19980211	
JP 2000509945	W	20000802	JP 98536142	A	19980211	200042
			WO 98DE377	A	19980211	

Priority Applications (No Type Date): DE 1007060 A 19970221

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9837716 A2 G 22 H04Q-007/22

Designated States (National): BR CN JP KR US

Designated States (Regional): AT BE CH DE DK ES FI FR GB GR IE IT LU MC
NL PT SE

EP 962106 A2 G H04Q-003/00 Based on patent WO 9837716
Designated States (Regional): AT BE DE ES FR GB IT
CN 1248377 A H04Q-003/00
BR 9807258 A H04Q-007/22 Based on patent WO 9837716
JP 2000509945 W 26 H04M-003/50 Based on patent WO 9837716
International Patent Class (Main): H04M-003/50; H04Q-003/00; H04Q-007/22
International Patent Class (Additional): H04M-003/42

Announcement provision method in communication network -

...Abstract (Basic): The method involves networked mobile switching centres and visitor location registers (MSC/VLR) to which **subscriber** access terminals (MS) can be connected. Announcement texts are introduced into a service control point (SCP). A message initiated on the basis of a **subscriber**'s call contains information on the supportability of announcements by an announcement unit (IP...

...message is received and evaluated before another message containing the announcement is transmitted. The announcement unit receiving a text converts it into an announcement for **transmission** over a **speech** channel to the caller...

...ADVANTAGE - Ensures only announcements are introduced into SCP. Ensures highly flexible system for implementing announcements by converting received **text** into **speech**.

...Title Terms: **NETWORK** ;

27/3,IC,K/10 (Item 10 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012019931

WPI Acc No: 1998-436841/199837

XRPX Acc No: N98-340382

Telecommunications system for deaf persons - has platform which routes call based on equipment type and which has signal detection circuitry detecting whether call is voice call

Patent Assignee: AT & T CORP (AMTT)

Inventor: AUGUST K G

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5787148	A	19980728	US 95583144	A	19951228	199837 B

Priority Applications (No Type Date): US 95583144 A 19951228

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
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US 5787148	A	8	H04M-011/00	
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International Patent Class (Main): H04M-011/00

International Patent Class (Additional): H04M-003/42; H04M-007/00

...Abstract (Basic): The system is for use in a telephone **network** to process communications with a telecommunications relay centre. The...

...destination for a text telephone party and information identifying the relay centre. The platform includes signal detection circuitry for determining that the call is a **voice** call. The platform **routes** the **voice** call to the relay centre, and the potential destination is identified to the relay centre in association with the voice call...

...ADVANTAGE - Allows **users** to have one telephone number for **text** and **voice** telephones...

27/3,IC,K/11 (Item 11 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

011648696

WPI Acc No: 1998-065604/199807

XRPX Acc No: N98-051614

Called number identity announcement e.g. for telephone system - involving calling party to receive voice announcement identifying called party prior to connection to called party allowing hangup

Patent Assignee: AT & T CORP (AMTT); AMERICAN TELEPHONE & TELEGRAPH CO (AMTT)

Inventor: SALIMANDO S C

Number of Countries: 027 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 818913	A2	19980114	EP 97111859	A	19970711	199807 B
JP 10084410	A	19980331	JP 97185630	A	19970711	199823
CA 2198797	A	19980112	CA 2198797	A	19970228	199927
US 5970133	A	19991019	US 96678933	A	19960712	199950
MX 9705116	A1	19980101	MX 975116	A	19970708	199952

Priority Applications (No Type Date): US 96678933 A 19960712

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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EP 818913	A2	E	13	H04M-003/50	
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Designated States (Regional): AL AT BE CH DE DK ES FI FR GB GR IE IT LI
LT LU LV MC NL PT RO SE SI

JP 10084410	A		11	H04M-001/56	
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CA 2198797	A			H04Q-003/72	
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US 5970133	A			H04M-003/42	
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MX 9705116	A1			H04M-001/00	
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International Patent Class (Main): H04M-001/00; H04M-001/56; H04M-003/42; H04M-003/50; H04Q-003/72

International Patent Class (Additional): H04M-001/57; H04M-011/00; H04Q-003/42; H04Q-003/545

...Abstract (Basic): The announcement system then converts **text** data to **voice** , or **passes** on **voice** data, and **delivers** it to the calling party before the connection is finalised. The message identifies the called party and allows time for the calling party to hang...

...ADVANTAGE - Allows **users** to ensure they have dialled correct number avoiding undesired charges and inefficient **network** use...

27/3,IC,K/12 (Item 12 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

010216031

WPI Acc No: 1995-117285/199516

XRPX Acc No: N95-092568

Producing and processing text documents - setting up text document using speech before converting to text data using speech detector and allowing text to be corrected, edited and extended by speech

Patent Assignee: ALCATEL SEL AG (COGE); ALCATEL NV (COGE)

Inventor: HUZENLAUB R; KOPP D; DE SANTIS G; RICCIO A; RIGOSI F
Number of Countries: 013 Number of Patents: 004
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 644680	A2	19950322	EP 94113016	A	19940820	199516 B
DE 4331710	A1	19950323	DE 4331710	A	19930917	199517
JP 7193647	A	19950728	JP 94221919	A	19940916	199539
US 5920835	A	19990706	US 94305849	A	19940914	199933
			US 97869476	A	19970605	

Priority Applications (No Type Date): DE 4331710 A 19930917

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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EP 644680	A2	G	12	H04M-003/42	
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Designated States (Regional): AT BE CH DE ES FR GB IT LI NL SE

DE 4331710	A1	11	H04M-003/50
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JP 7193647	A	7	H04M-011/00
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US 5920835	A		G10L-005/06	Cont of application US 94305849
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International Patent Class (Main): G10L-005/06; H04M-003/42; H04M-003/50; H04M-011/00

International Patent Class (Additional): G06F-003/16; G06F-013/00; G10L-003/00; G10L-005/02; G10L-007/08; H04M-011/10; H04N-001/00

... **setting up** text document using speech before converting to text data using speech detector and allowing text to be corrected, edited and extended by speech

...Abstract (Basic): text documents to be dictated and transmitted using a telecommunication device. Text is dictated in the form of speech. The speech is then converted to **text** data using **speech** recognition. The text data can be corrected by means of speech and can be edited in text data. The text can be transmitted as text data to **subscribers** via a telecommunication **network** .

...

...The device for dictating and transmitting text documents includes a dictating machine (HS). A device is provided for **transmitting** spoken **speech** to a **speech** detector (SEK). The detector (SEK) converts speech into text data. Software is provided to correct (i) the **text** data using **speech** . Software is also provided to edit (ii) the text data. Another device transmits the text data to a further **subscriber** .

27/3,IC,K/13 (Item 13 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

008798331

WPI Acc No: 1991-302345/199141

XRPX Acc No: N91-231582

Network **order entry service for telecommunications system** - can receive **orders by facsimile, transforming data into text form using OCR circuitry, and stores text converted speech**

Patent Assignee: ANONYMOUS (ANON)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
TP 99105	A	19910925	TP 9199105	A	19910920	199141 B

Priority Applications (No Type Date): TP 9199105 A 19910920

International Patent Class (Additional): H04M-000/01

Network order entry service for telecommunications system...

...can receive orders by facsimile, transforming data into text form using OCR circuitry, and stores text converted speech

...Abstract (Basic): An automated Order Entry System (OES) resides in a telecommunications **network** and is arranged to receive information from callers desiring to place orders with a called party (**subscriber**). The information may be entered by callers as speech and/or as touch tone digits, in response to voice prompts generated by, for example, an AT and T Conversant Voice Response System located in the **network**. Information entered by callers in speech form is processed by speech-to-text conversion circuitry and stored in an electronic mail-box assigned to the **subscriber** or combined with other orders, possibly converted to electronic data interchange format, and forwarded to the **subscriber**'s computer. The OES can also receive orders by FAX, transform the information to text form using optical character recognition circuitry, and combine the FAX orders with **speech**-based orders before being **transmitted** to the **subscriber**. (Dwg.No.0/0)

Title Terms: **NETWORK** ;

27/3,IC,K/14 (Item 14 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

007836207

WPI Acc No: 1989-101319/198914

XRPX Acc No: N89-077303

Multi-media mail system consolidating voice and text mail - has transmit-receive mode selectors between analog telephone network and paired voice and text mail centres

Patent Assignee: HITACHI LTD (HITA)

Inventor: SHIBATA Y

Number of Countries: 007 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 309993	A	19890405	EP 88115888	A	19880927	198914 B
JP 1086643	A	19890331	JP 87242438	A	19870929	198919
US 4972462	A	19901120	US 88249714	A	19880927	199049
EP 309993	B1	19950503	EP 88115888	A	19880927	199522
DE 3853707	G	19950608	DE 3853707	A	19880927	199528
			EP 88115888	A	19880927	

Priority Applications (No Type Date): JP 87242438 A 19870929

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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EP 309993	A	E	19		
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Designated States (Regional): DE FR GB IT NL

EP 309993	B1	E	20	H04M-003/50	
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Designated States (Regional): DE FR GB IT NL

DE 3853707	G			H04M-003/50	Based on patent EP 309993
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International Patent Class (Main): H04M-003/50

International Patent Class (Additional): H04L-011/20; H04L-012/54;

H04M-011/00; H04Q-003/00

... has transmit-receive mode selectors between analog telephone network and paired voice and text mail centres

- ...Abstract (Basic): The system has a voice mail system and a text mail system utilising an analog telephone **network** (1004). A centre data/**voice transmit** /receive mode selector (1003) is provided between a paired voice mail centre (1002) and text mail centre (1000) and the analog telephone **network** . A terminal data/**voice transmit** / receive mode selector (1003) is provided between a paired voice mail terminal (1007) and text mail terminal (1006) and the analog telephone **network** .
- ...
- ...Text mail centre and voice mail centre are physically in one centre but are logically or functionally separated. **Subscriber** data, charge data, and voice mail and text mail control information are communicated between a text mail centre processor and a voice mail centre processor. When turn-off of a modem carrier is detected, a data/**voice transmit** /receiver mode selector provided at a predetermined section of the system selects a **voice transmitter** /receiver (**voice** mail centre, and microphone and speaker of the terminal). When the modem carrier is detected and a predetermined specific data is also detected, the selector
- ...Abstract (Equivalent): A multimedia mail system having a voice mail system and a text mail system utilizing an analog telephone **network** (1004), comprising: a voice mail centre (1002) and a text mail centre (1001), and a centre data/**voice transmit** /receive mode selector (1003) provided between the voice mail centre and text mail centre, and said analog telephone **network** (1004); characterised by said centre data/**voice transmit** /receive mode selector (1003) being adapted to freely switch **text** data and **voice** data into one communication, whereby for switching voice data to text data a carrier detect signal (CD) and a data indicative of this switching is used; a terminal data/**voice transmit** /receive mode selector (1003') provided between a voice mail terminal (1007) and a text mail terminal (1006) and said analog telephone **network** ; and the voice mail terminal (1007) and text mail terminal (1006) constituting a multimedia terminal being capable of sending and/or receiving voice data and...
- ...Abstract (Equivalent): A multimedia mail system utilises an analog telephone **network** and interconnects processors at a voice main centre and a text mail centre and provides data/**voice transmit** /receive mode selectors between the analog telephone **network** and the paired voice mail centre and text mail centre and between the analog telephone **network** and paired voice mail terminal and text mail terminal so that voice and text data can be switched during communication to provide a consolidated voice...
- ...Title Terms: **NETWORK** ;

27/3,IC,K/15 (Item 1 from file: 347)

DIALOG(R)File 347:JAPIO

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05404667

PATIENT INFORMATION SYSTEM

PUB. NO.: 09-019467 [JP 9019467 A]

PUBLISHED: January 21, 1997 (19970121)

INVENTOR(s): SAKUSHIMA HIROMI

YAMAOKA MEGUMI

APPLICANT(s): MATSUSHITA ELECTRIC IND CO LTD [000582] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 07-169466 [JP 95169466]

FILED: July 05, 1995 (19950705)

INTL CLASS: [6] A61G-012/00; H04M-011/08

ABSTRACT

...SOLVED: To quickly transfer a message between a remote place and a nurse station, efficiently provide information for a corresponding patient, and smoothly perform a **text /voice** mixed information **transmission** through a document such as chart...

...SOLUTION: A portable radiocommunication slave machine 133 performs radiocommunication with a radiocommunication parent machine 132, and is connected to a local area **network** through a **network** connecting device 130 having slave **user** control means 131. A nursing information system **server** 110 is further connected to the local area **network**. An input device 120 of the nursing information system **server** 110 is provided with a keyboard 121, a mouse 122, and a microphone 123, and an output device 124 thereof is provided with a display...
?

?t /3,ic,k/1-10

31/3,IC,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

012943677

WPI Acc No: 2000-115530/200010

XRPX Acc No: N00-087402

**Interactive prosody user interface in text-to- speech system,
speech synthesizer system**

Patent Assignee: LUCENT TECHNOLOGIES INC (LUCE)

Inventor: TANENBLATT M A

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 6006187	A	19991221	US 96720759	A	19961001	200010 B

Priority Applications (No Type Date): US 96720759 A 19961001

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 6006187	A	11	G10L-005/02		

International Patent Class (Main): G10L-005/02

**Interactive prosody user interface in text-to- speech system,
speech synthesizer system**

Abstract (Basic):

... **Duration** controller sets speaking rate relative word **duration** of selected words to be uttered by **synthesized** voice. A creation unit forms text string using selected words and **prosody** characteristic, to apply changed **prosody** characteristic to voiced output of at least one of displayed words as to which changed **prosody** characteristic is effected.

... The **duration** controller enables **user** to dynamically effect change in **prosody** characteristic for one of displayed words. Words and punctuation in text input into word boxes is selected using mouse click, after which it is displayed visually. The **duration** controller operates in conjunction with the display unit which has indicia of change in one **prosody** characteristic for the displayed words. An INDEPENDENT CLAIM is also included for the altering method of **prosody** characteristics of **synthesized** voice in **text -to-speech** system...

...In **text -to-speech** system, speech **synthesizer** system for controlling acoustical characteristic of **synthesized** voice...

...The **prosody** **user** interface includes unlimited undo feature which allows any changes that are made to be reversed, thus giving the **user** freedom to explore various alternatives while retaining the ability to return to the previous state...

...The figure illustrates the flowchart for transmitting escape sequences relating to phrase contours to **text -to-speech synthesizer** process

...

...Title Terms: **PROSODY ; USER ;**

31/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012913566

WPI Acc No: 2000-085402/200007

XRPX Acc No: N00-066931

Integrated messaging and voice-free cellular telephone communication system for use by hearing impaired, mute and deaf person

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)

Inventor: BRUNET P T; ITTYCHERIAH A P; NARAYANASWAMI C; PICHENY M A; RAMABHADHAN B

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5995590	A	19991130	US 9835493	A	19980305	200007 B

Priority Applications (No Type Date): US 9835493 A 19980305

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5995590	A	7	H04M-011/00	

International Patent Class (Main): H04M-011/00

Abstract (Basic):

... **Text** to **speech** converter (14) connected to text data input device (12) of telephone set, converts text of message into **synthesized** speech signals. Input key of device (12) represents separate entire group of selected words and phrases. Memory of converter (14) stores words and phrases in form of **synthesized** speech signals. Speech to text converter of other telephone set is connected through link.

... The speech to text converter converts the speech signal to text signals in response to speech signals from the **text** to **speech** converter of other telephone set...

...provides immediate and interactive response. To simplify the task of typing or writing with input device, several preselected words or phrases are used by the **user** , thereby avoids guide person for deaf, mute and hearing impaired person. Exhibits automatic answering function when the hearing impaired person does not take the call...

...and reconfigurable, thereby shorthand notation is facilitated and amount of typing is reduced and these techniques allow for more interactivity during call and also reduces **duration** of call...

...**Text** to **speech** converter (14

31/3,IC,K/3 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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010107711

WPI Acc No: 1995-008964/199502

XRPX Acc No: N95-007432

Message broadcasting unit in radio paging system - transmits message to specific users by means of coded message system and generates analog audio waveform

Patent Assignee: IBM CORP (IBMC); INT BUSINESS MACHINES CORP (IBMC)

Inventor: LEMAIRE C A; STRIEMER B L

Number of Countries: 002 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 6237207	A	19940823	JP 93315254	A	19931215	199502 B
US 5594658	A	19970114	US 92993278	A	19921218	199709
			US 95469307	A	19950606	

US 5613038 A 19970318 US 92993278 A 19921218 199717

Priority Applications (No Type Date): US 92993278 A 19921218; US 95469307 A 19950606

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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JP 6237207	A		9	H04B-007/26	
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US 5594658	A		8	G06F-017/00	Div ex application US 92993278
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US 5613038	A		8	G10L-005/02	
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International Patent Class (Main): G06F-017/00; G10L-005/02; H04B-007/26

International Patent Class (Additional): G10L-009/00; H04M-001/64

... **transmits message to specific users by means of coded message system and generates analog audio waveform**

...Abstract (Basic): address (143) of the specific receiver. The receivers receive the transmitted message pattern using a receiver antenna and store it in a data buffer. The **user** operates a switch control unit in the receiver for choosing the stored data by means of mode control buttons (157...

...program memory. When the message and receiver addresses coincides, a selector selects one message in the text portion of messages. The corresponding voice waveform is **generated** by the voice processor for the selected message. The analog output from the voice processor is amplified by an amplifier and fed to a speaker...

...USE/ADVANTAGE - Digital paging system. Facilitates individual transmission of messages according to **user** demand...

...Abstract (Equivalent): switch means operable by a **user** of said portable communications receiver for choosing one message among said stored selected messages, and wherein said switch means includes...

...a first switch for sending a current one of said messages in said sequence to a **text -to-speech** conversion means, wherein said **text -to-speech** conversion means is coupled to said storing means and responsive to said switch means, for producing analog speech waveforms directly corresponding to the text portion...

...for choosing a next message in said sequence as said current message, and wherein another operation of said second switch increases the speed of said **text -to-speech** conversion means...

...A communications system for transmitting multiple individually addressed messages to a large number of **users** at different locations, comprising...

...a first switch operable by a **user** for choosing a current one of said messages...

...a second switch operable by said **user** for choosing a previous one of said messages...

...a third switch operable by said **user** for choosing a next one of said messages...

...**text -to-speech** conversion means responsive to said switch means and coupled to said data storage means for **generating** analog speech waveforms directly representing the text portion of said chosen message

...Title Terms: **USER** ;

31/3,IC,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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009526184
WPI Acc No: 1993-219725/199327
XRPX Acc No: N93-168408

**Computer graphical message box location method for blind person -
generating while noise when pointer is on message box but not on button
and using test-to-speech system for keystroke announcements**

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC)

Inventor: MCKIEL F A

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5223828	A	19930629	US 91746838	A	19910819	199327 B

Priority Applications (No Type Date): US 91746838 A 19910819

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5223828	A	9	H04Q-001/00	

International Patent Class (Main): H04Q-001/00

... generating while noise when pointer is on message box but not on
button and using test-to-speech system for keystroke announcements

...Abstract (Basic): When a message box first appears, the text contents
are announced using a **text -to-speech** system. After the text is
announced, the push buttons available to respond to or cancel the
message box are also announced in order from left to right. Next, a
homing singla is provided for finding the message box. The homing
signla is a tone that increases in **pitch** as the pointer approaches
the message box. When the pointer enters the message box, the message
box text and the available push buttons are reannounced...

...As long as the pointer is on a button, the system remains silent. If the
user desires to select a push button other than the default, the
user may move the pointer to the left toward the other buttons...

...USE/ADVANTAGE - Allows blind person to access and use computer graphical
user interface...

...Title Terms: **GENERATE** ;

31/3,IC,K/5 (Item 5 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

008882986
WPI Acc No: 1992-010255/199202
XRPX Acc No: N92-007875

**Text to speech converter for handicapped users - times input to
synthesiser with natural speech rhythm by rules identifying terms and
recognising syntactic information**

Patent Assignee: AMERICAN TELEPHONE & TELEGRAPH CO (AMTT); AT & T CORP
(AMTT); AT & T BELL LAB (AMTT)

Inventor: BACHENKO J C

Number of Countries: 005 Number of Patents: 007

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
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EP 465058	A	19920108	EP 91305601	A	19910620	199202 B
CA 2043667	A	19911229				199213
US 5157759	A	19921020	US 90546127	A	19900628	199245
EP 465058	A3	19950322	EP 91305601	A	19910620	199543
CA 2043667	C	19960213	CA 2043667	A	19910531	199617
EP 465058	B1	19990825	EP 91305601	A	19910620	199939
DE 69131549	E	19990930	DE 631549	A	19910620	199946
			EP 91305601	A	19910620	

Priority Applications (No Type Date): US 90546127 A 19900628

Patent Details:

Patent No	Kind	Lan	Pg	Main	IPC	Filing	Notes
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EP 465058	A		14				
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Designated States (Regional): DE FR GB

US 5157759	A		9	G10L-009/00			
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EP 465058	B1	E		G10L-005/06			
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Designated States (Regional): DE FR GB

DE 69131549	E			G10L-005/06	Based on patent	EP 465058	
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CA 2043667	C			G10L-009/08			
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International Patent Class (Main): G10L-005/06; G10L-009/00; G10L-009/08

International Patent Class (Additional): G09B-021/00; G10L-005/02

Text to speech converter for handicapped users - ...

...times input to synthesiser with natural speech rhythm by rules
identifying terms and recognising syntactic information

...Abstract (Basic): USE/ADVANTAGE - By deaf persons or sufferers from
speech impediments. Freely **generated** text sequence is synthesised
with proper emphases and pauses, without intervention of attendant.
(14pp Dwg.No.1/4)

...Abstract (Equivalent): The converter for synthesising a speech signal
has a word detector responsive to a freely **generated** text signal for
detecting individual words in the text signal and developing a string
of words to be synthesised. A categorising device analyses each word...

...A syntax augmenting device considers each word in the string and inserts
a pause **generation** signal in the string of words, before or after the
considered word, when appropriate, based on the category of the
considered word. The syntax augmenting device inserts the pause
generation signal before or after the considered word when
appropriate, based on the considered word's category and the category
of the one of the words...

...Title Terms: **USER** ;

31/3,IC,K/6 (Item 1 from file: 347)

DIALOG(R)File 347:JAPIO

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05888385

VOICE **SYNTHESIZER**

PUB. NO.: 10-171485 [JP 10171485 A]

PUBLISHED: June 26, 1998 (19980626)

INVENTOR(s): YAMAGAMI KATSUYOSHI

MATSUI KENJI

APPLICANT(s): MATSUSHITA ELECTRIC IND CO LTD [000582] (A Japanese Company
or Corporation), JP (Japan)

APPL. NO.: 08-331817 [JP 96331817]

FILED: December 12, 1996 (19961212)

INTL CLASS: [6] G10L-003/00; G06F-017/28; G10L-005/04

VOICE SYNTHESIZER

ABSTRACT

PROBLEM TO BE SOLVED: To provide a voice **synthesizer** presenting **text** information in **voice** comprehensible to each **user** .

...
... inserts an important part referring to an important part pattern table 104; inserts a control command in the acoustic parameter based on those results; a **prosody** information **generation** part 106 **generates prosody** information, an acoustic parameter; and an acoustic processing part 107 outputs vocally. Moreover, a **user** is identified in a **user** -identification part 108, and the contents of the change processing of the parsing result by the parsing result changing part 105 is controlled according to the **user** .

31/3,IC,K/7 (Item 2 from file: 347)

DIALOG(R)File 347:JAPIO

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05727983

TEXT VOICE CONVERTING DEVICE

PUB. NO.: 10-011083 [JP 10011083 A]

PUBLISHED: January 16, 1998 (19980116)

INVENTOR(s): TSUKAMOTO KAORU

APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 08-162886 [JP 96162886]

FILED: June 24, 1996 (19960624)

INTL CLASS: [6] G10L-003/00; G06F-017/21; G10L-005/04

TEXT VOICE CONVERTING DEVICE

ABSTRACT

PROBLEM TO BE SOLVED: To provide the **text voice** converting device in which **synthesized** sounds of many kinds of pronunciation styles are **generated** and the reading is conducted with the phoneme patterns matched with the liking of a **user** .

...
...SOLUTION: A synthesis parameter **generating** section 13 takes out the corresponding voice piece data based on a phoneme symbol column from a voice piece data storage section 14 and **generates** voice synthesis rhythm parameters such as the **duration** of phonemes, the length of a pause, power and fundamental frequency patterns. An uttering style specifying section 17 specifies one desired uttering style from plural...

... styles covering a reading style to a conversation style. A synthesis parameter changing means 16 deforms the voice synthesis phoneme parameters in accordance with the **user** 's specification made by the section 17. A voice synthesis section 15 **synthesizes** voices and outputs them in accordance with the voice synthesis phoneme parameters.

31/3,IC,K/8 (Item 3 from file: 347)

DIALOG(R)File 347:JAPIO

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04951896

TEXT RECITATION DEVICE

PUB. NO.: 07-244496 [JP 7244496 A]
PUBLISHED: September 19, 1995 (19950919)
INVENTOR(s): NIIMURA TAKAHIKO
APPLICANT(s): N T T DATA TSUSHIN KK [000000] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 06-036190 [JP 9436190]
FILED: March 07, 1994 (19940307)
INTL CLASS: [6] G10L-005/04; G06F-003/16; G10L-003/00

ABSTRACT

PURPOSE: To provide the text recitation device which **generates** a **natural** sensational **speech** where tastes of individual device **users** are reflected...

...CONSTITUTION: On the basis of rhythm parameters of a calm speech which are **generated** by the rhythms of an input **text**, ideal sensational **speech** rhythm parameters showing a specific feeling are **generated** from relative value information. An element piece selection part 110 selects and extracts the element pieces of the rhythm parameters which are closest to the...

... the element pieces within a range wherein naturalness is held and puts them close to the feeling speech rhythm parameters to obtain a desired feeling **synthesized** speech.

31/3,IC,K/9 (Item 4 from file: 347)

DIALOG(R)File 347:JAPIO

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03613294

TEXT SOUND CONVERTER

PUB. NO.: 03-276194 [JP 3276194 A]
PUBLISHED: December 06, 1991 (19911206)
INVENTOR(s): YATO TAKASHI
APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 02-078338 [JP 9078338]
FILED: March 27, 1990 (19900327)
INTL CLASS: [5] G10L-003/00
JOURNAL: Section: P, Section No. 1323, Vol. 16, No. 100, Pg. 16, March 11, 1992 (19920311)

TEXT SOUND CONVERTER

ABSTRACT

PURPOSE: To obtain a desired **synthesized** tone equipped originally with reading, accent, **intonation** and breath, etc., which are originally controlled by a **user**, with simple configuration by providing mode set and control parts...

...CONSTITUTION: When the mode set part 40 selects any one of modes such as a **text / sound** conversion mode, phoneme sound and meter symbol train output mode and phoneme sound and meter symbol train **synthesizing** mode according to a designation from an external part, the control part 50 discriminates the mode and controls the input/output of a text and a phoneme sound and meter symbol train. When the **text / sound** conversion mode is set, the control part 50 inputs the text and a text analysis part

30 analyzes the text. Then, the result of the analysis is imparted through a sound **synthesizing** part 60 to a loudspeaker 61. When the phoneme sound and meter symbol train output mode is set, the control part 50 analyzes the text at the text analysis part 30 and outputs the **generated** phoneme sound and meter symbol train in the form of a character code to the external part. When the phoneme sound and meter symbol train **synthesizing** mode is set, the control part 50 directly outputs the phoneme sound and meter symbol train inputted from the external part, through the sound **synthesizing** part 60. Thus, the **user** can freely change the phoneme sound and meter symbol train and easily obtain the desired **synthesized** tone.

31/3,IC,K/10 (Item 5 from file: 347)

DIALOG(R)File 347:JAPIO

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03613293

TEXT SOUND CONVERTER

PUB. NO.: 03-276193 [JP 3276193 A]
PUBLISHED: December 06, 1991 (19911206)
INVENTOR(s): YATO TAKASHI
APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 02-078337 [JP 9078337]
FILED: March 27, 1990 (19900327)
INTL CLASS: [5] G10L-003/00; G06F-015/20
JOURNAL: Section: P, Section No. 1323, Vol. 16, No. 100, Pg. 16, March 11, 1992 (19920311)

TEXT SOUND CONVERTER

ABSTRACT

PURPOSE: To obtain a **synthesized** sound desired for a **user** with simple and easy operations by providing a means to control a phoneme sound and meter symbol train **generating** means...

... auxiliary memory 23. When there is no symbol to show text analysis supporting information, the text is analyzed by the phoneme sound and meter symbol **generating** means 50. Then, a phoneme sound and meter symbol train required for reading a sentence as a sound is **generated** through a word division processing means 51, read processing means 52, accent application imparting means 53, pause and **intonation** setting means 54, and in a sound **synthesizing** part 60, the sound corresponding to the input text is **synthesized** and outputted from a loudspeaker 61. When the input text includes the symbol to show the text analysis supporting information, the designation of the symbol...

...sentence can be read as intended. In the case of applying only a desired support element to be designated, the other element is automatically **generated** by the phoneme sound and meter symbol train **generating** means 50.

?

?t /3,ic,k/1-17

36/3,IC,K/1 (Item 1 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

013229834

WPI Acc No: 2000-401708/200035

XRPX Acc No: N00-300848

Extracting formant-based source signals and filter parameters from speech signal by extracting z-plane complex log of residual signal complex spectrum

Patent Assignee: MATSUSHITA ELECTRIC IND CO LTD (MATU); MATSUSHITA DENKI SANGYO KK (MATU)

Inventor: PEARSON S

Number of Countries: 026 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 1005021	A2	20000531	EP 99309294	A	19991122	200035 B
JP 2000231394	A	20000822	JP 99332612	A	19991124	200045

Priority Applications (No Type Date): US 98200335 A 19981125

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

EP 1005021 A2 E 16 G10L-019/06

Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT

LI LT LU LV MC MK NL PT RO SE SI

JP 2000231394 A 48 G10L-013/00

International Patent Class (Main): G10L-013/00; G10L-019/06

International Patent Class (Additional): G10L-013/04

Abstract (Basic):

... Method consists in defining a filter model (12) to produce a filter (10), applying the speech signal to the filter to **generate** a residual signal, processing this by extracting time domain data to extract a set of data points defining a line of segments, calculating the length...

...parameter. The steps are repeated (16) until the cost parameter is minimized. A second filter inverse to the first processes the extracted source signal to **generate synthesized** speech

... Method is for use in constructing **text -to-speech** and music **synthesizers** and speech coding systems...

...Method produces a **natural sounding waveform** without distortions due to discontinuities...

36/3,IC,K/2 (Item 2 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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012671893

WPI Acc No: 1999-478000/199940

XRPX Acc No: N99-355782

Parametric synthetic text-to- speech generating method for percussive musical instrument e.g. plucked violin

Patent Assignee: APPLE COMPUTER INC (APPY)

Inventor: CECYS M L

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5930755	A	19990727	US 94212602	A	19940311	199940 B
			US 97779424	A	19970107	

Priority Applications (No Type Date): US 94212602 A 19940311; US 97779424 A 19970107

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
US 5930755	A	16	G10L-005/02	Cont of application US 94212602

International Patent Class (Main): G10L-005/02

Parametric synthetic text-to- speech generating method for percussive musical instrument e.g. plucked violin

Abstract (Basic):

... A set of **synthesizer** control parameters representative of text to be spoken, is **generated** and recorded. Among the recorded **sound samples** , a **voice** source is selected. Based on the selected voice source, speech **synthesizer** control parameters are converted into output **waveforms** representative of synthetic speech to be spoken.

... An INDEPENDENT CLAIM is also included for the parametric synthetic system for the **text -to- speech** conversion...

...For **generating** parametric synthetic **text -to-speech** used in non-human sound sources like electronic systems, talking teakettle, animal and percussive musical instrument e.g. snare drum, plucked violin...

...In the synthetic **text -to-speech** **generation** , the output **waveforms** representative of synthetic speech, can be provided by selecting atleast one voice source in a speech **synthesizer** .

...

...The figure shows the sub-segments of recorded **sound sample** used in **text -to-speech** conversion

...Title Terms: **GENERATE** ;

36/3,IC,K/3 (Item 3 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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011955584

WPI Acc No: 1998-372494/199832

XRPX Acc No: N98-292139

Text **speech-synthesis apparatus for FM data multiplex broadcasting, VICS - has control unit that performs speech synthesis or rule synthesis depending on correspondence or non-correspondence of word identification attribute row and example pattern**

Patent Assignee: FUJITSU TEN LTD (FUTE)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 10149188	A	19980602	JP 96310890	A	19961121	199832 B

Priority Applications (No Type Date): JP 96310890 A 19961121

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 10149188	A	5	G10L-003/00	

International Patent Class (Main): G10L-003/00
International Patent Class (Additional): G10L-005/02

Text speech-synthesis apparatus for FM data multiplex broadcasting,
VICS...

...Abstract (Basic): if the attribute row from the analyser corresponds with the example pattern, and performs speech synthesis if agreement is obtained. Otherwise, a speech pattern is **generated**, and the rule synthesis is performed in the **intonation** of the phonogram row using a **pitch** pattern...

...The speech synthesis is performed using the **intonation** peculiar to connection words linking words to form one sentence. An example table stores the fitting example pattern consisting of the **intonation** used for speech synthesis. A fitting type rhythm **generator** forms the **pitch** pattern from the example pattern, and uses the **pitch** pattern to link the **waveform** of the audio unit of the phonogram row of the word row...

36/3,IC,K/4 (Item 4 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

011754279

WPI Acc No: 1998-171189/199816

XRPX Acc No: N98-136025

Text speech synthesis method - setting prosodic information for
phoneme sequence of each word of word sequence obtained by analysis of
input text by referring to word dictionary with speech waveform
sequence obtained from phoneme sequence of each word

Patent Assignee: NIPPON TELEGRAPH & TELEPHONE CORP (NITE)

Inventor: ABE M

Number of Countries: 025 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 831460	A2	19980325	EP 97116540	A	19970923	199816 B
JP 10153998	A	19980609	JP 97239775	A	19970904	199833
US 5940797	A	19990817	US 97933140	A	19970918	199939

Priority Applications (No Type Date): JP 97239775 A 19970904; JP 96251707 A 19960924

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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EP 831460	A2	E	13	G10L-005/04	
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Designated States (Regional): AL AT BE CH DE DK ES FI FR GB GR IE IT LI

LT LU LV MC NL PT RO SE SI

JP 10153998	A		10	G10L-003/00	
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US 5940797	A			G10L-005/02	
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International Patent Class (Main): G10L-003/00; G10L-005/02; G10L-005/04

Text speech synthesis method...

...setting prosodic information for phoneme sequence of each word of word sequence obtained by analysis of input text by referring to word dictionary with speech waveform sequence obtained from phoneme sequence of each word

...Abstract (Basic): reference to a word dictionary and identifying a sequence of words in the input text to obtain a sequence of phonemes of each word. A **prosodic** information on the phonemes is set in each word. Phoneme **waveforms** are selected from a speech **waveform**

dictionary which corresponds to the phonemes in each word to **generate** a sequence of phoneme **waveforms** .

...

...A **prosodic** information is extracted from input actual speech. One part of the extracted **prosodic** information and one part of the set **prosodic** information is selected. A synthesised speech is **generated** by controlling the sequence of phoneme **waveforms** with the selected **prosodic** information

...Title Terms: **WAVEFORM** ;

36/3,IC,K/5 (Item 5 from file: 350)
DIALOG(R) File 350:Derwent WPIX
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011659818

WPI Acc No: 1998-076726/199807

XRPX Acc No: N98-061382

Synthetic text-to- speech generating - converts speech synthesiser control parameters into output wave forms representative of synthetic speech to be spoken by selecting and combining at least two voice sources

Patent Assignee: APPLE COMPUTER INC (APPY)

Inventor: CECYS M L

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5704007	A	19971230	US 94212488	A	19940311	199807 B
			US 96727845	A	19961004	

Priority Applications (No Type Date): US 94212488 A 19940311; US 96727845 A 19961004

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

US 5704007 A 17 G10L-005/02 Cont of application US 94212488

International Patent Class (Main): G10L-005/02

International Patent Class (Additional): G10L-009/00

Synthetic text-to- speech generating - ...

...converts speech synthesiser control parameters into output wave forms representative of synthetic speech to be spoken by selecting and combining at least two voice sources

...Abstract (Basic): The method involves **generating** a set of speech synthesiser control parameters representative of text to be spoken, and converting the speech synthesiser control parameters into output **wave forms** . The latter is representative of the synthetic speech to be spoken by selecting and combining at least two voice sources from a number of voice sources in a speech synthesiser. That **generates** a combined voice source and by passing the combined voice source through an acoustic model of a human vocal tract...

...The number of voice sources has spectral content, which most closely matches that of the **generated** set of speech synthesiser control parameters and includes a normal voice source and a bright voice source voice source, representative of text to be spoken. The speech synthesiser control parameters are converted into output **wave forms** representative of the synthetic speech to be spoken by selecting and combining at least two voice sources from the number of voice sources in a speech synthesiser to **generate** a combined voice source...

...ADVANTAGE - Provides multiple voice source, each of which has certain desirable spectral content such that more **natural** human like synthesised **speech** can be **generated** with reduced reliance on signal processing...
...Title Terms: **GENERATE** ;

36/3,IC,K/6 (Item 6 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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011469438

WPI Acc No: 1997-447345/199741

XRPX Acc No: N97-372821

Mandarin syllable-signal synthesis method - synthesising periodical waveform part by performing time proportionated-interpolation and resampling operation

Patent Assignee: GUU H (GUUH-I)

Inventor: GUU H

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
TW 309588	A	19970701	TW 96116039	A	19961224	199741 B

Priority Applications (No Type Date): TW 96116039 A 19961224

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
TW 309588	A	28	G10L-009/00	

International Patent Class (Main): G10L-009/00

... **synthesising periodical waveform part by performing time proportionated-interpolation and resampling operation**

...Abstract (Basic): The method is based on time-domain **waveform** processing. The effect of non-linearly warping the formant trace is largely decreased when changing one of the values of the two parameters, **duration** and **pitch** -frequency trace. The method **synthesizes** the periodical-**waveform** part by performing a type of time-proportionated-interpolation and a type of resampling operation. This lets the flexibility of independent control of the three factors, **duration** , **pitch** -frequency trace, and vocal-track length, be largely increased. Among the three, the factor of vocal-track length is new...

...When the values of the two factors, vocal-track length and **pitch** -frequency trace's height, are appropriately set, many distinct timbres can be **synthesized** by manipulating only a male's original syllable **waveforms** , e.g. the timbres of cartoon actors, children, women, and men...

...USE/ADVANTAGE - For implementing prototype **text -to-speech** system which can utter sentences, in real-time, in the timbre specified by control messages within input text. For synthesis of dialogues of dramas. Has increased flexibility in independent control of parameters and capability to **generate** many timbres...

...Title Terms: **WAVEFORM** ;

36/3,IC,K/7 (Item 7 from file: 350)
DIALOG(R)File 350:Derwent WPIX
(c) 2000 Derwent Info Ltd. All rts. reserv.

011087670

WPI Acc No: 1997-065594/199706

XRPX Acc No: N97-053924

Speech synthesiser - converts input text to sequence of representations of syllables or other phonetic units and retrieves stored parts of data to generate corresp. waveforms, and defines constant duration for regular beat period

Patent Assignee: BRITISH TELECOM PLC (BRTE)

Inventor: BREEN A P

Number of Countries: 071 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9642079	A1	19961227	WO 96GB1430	A	19960613	199706 B
AU 9662311	A	19970109	AU 9662311	A	19960613	199717
EP 832481	A1	19980401	EP 96920927	A	19960613	199817
			WO 96GB1430	A	19960613	
JP 11507740	W	19990706	WO 96GB1430	A	19960613	199937
			JP 97502810	A	19960613	
AU 713208	B	19991125	AU 9662311	A	19960613	200006

Priority Applications (No Type Date): EP 95304079 A 19950613

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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WO 9642079	A1	E	12	G10L-005/04	
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Designated States (National): AL AM AT AU AZ BB BG BR BY CA CH CN CZ DE DK EE ES FI GB GE HU IL IS JP KE KG KP KR KZ LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK TJ TM TR TT UA UG US UZ VN

Designated States (Regional): AT BE CH DE DK EA ES FI FR GB GR IE IT KE LS LU MC MW NL OA PT SD SE SZ UG

AU 713208	B			G10L-005/04	Previous Publ. patent AU 9662311
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Based on patent WO 9642079

AU 9662311	A			G10L-005/04	Based on patent WO 9642079
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EP 832481	A1	E		G10L-005/04	Based on patent WO 9642079
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Designated States (Regional): BE DE FR GB IT

JP 11507740	W		14	G10L-003/00	Based on patent WO 9642079
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International Patent Class (Main): G10L-003/00; G10L-005/04

International Patent Class (Additional): G10L-005/02

... converts input text to sequence of representations of syllables or other phonetic units and retrieves stored parts of data to generate corresp. waveforms, and defines constant duration for regular beat period

...Abstract (Basic): The speech synthesiser has a device for supplying a sequence of representations of phonetic units, and a device for retrieving stored portions of data to **generate waveforms** corresponding to the phonetic units. A device determines the durations for the phonetic units, and a processing device processes and adjusts the durations of the **waveforms** according to the determined durations

...The determiner is operable to define a constant **duration** corresponding to a regular beat period and adjusts the **duration** depending on the nature of the phonetic unit and/or its context within the sequence. The device identifies word grouping in the sequence, and the...

...USE/ADVANTAGE - E.g. for **text -to-speech** synthesisers...

...Title Terms: **GENERATE** ;

36/3,IC,K/8 (Item 8 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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010997609

WPI Acc No: 1996-494558/199649

XRFX Acc No: N96-417079

Audio synthesiser for text speech synthesis - has waveform super position processing part that produces source signal of audio data that drives vocal tract filter part

Patent Assignee: TOSHIBA KK (TOKE)

Inventor: AKAMINE M; KAGOSHIMA T

Number of Countries: 002 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 8254993	A	19961001	JP 9557773	A	19950316	199649 B
US 5890118	A	19990330	US 96613093	A	19960308	199920

Priority Applications (No Type Date): JP 9557773 A 19950316

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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JP 8254993	A	11	G10L-005/04		
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US 5890118	A		G10L-009/04		
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International Patent Class (Main): G10L-005/04; G10L-009/04

International Patent Class (Additional): G10L-009/10

Audio synthesiser for text speech synthesis...

...has waveform super position processing part that produces source signal of audio data that drives vocal tract filter part

...Abstract (Basic): The synthesiser comprises a memory unit (21), which outputs selected **waveforms** from stored **waveforms** representing frames of source signals of audio data on passing information corresponding to the audio signal which is to be **synthesized**. The selected **waveforms** are interpolated by an interpolating unit (22). Corresponding to two continuous outputs from the memory unit, which results in a source signal **waveform** of an audio data...

...The source signal **waveforms** are subjected to superposition in the positions determined by a position determining unit (11). Superposition of the source signal **waveform** is carried out by a superposition processing unit (23) is whose output drives a vocal tract filter (15). The vocal tract filter approximates the vocal...

...USE/ADVANTAGE - For producing composite tone audio from informations like tone symbol string, **pitch** and tone continuation time length. Reduces variation in tone and **pitch**, thus providing smooth natural continuous composite tone...

...Title Terms: **WAVEFORM** ;

36/3,IC,K/9 (Item 9 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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010684335

WPI Acc No: 1996-181291/199619

XRFX Acc No: N96-152330

Speech synthesis method using concatenation and partial overlapping of waveforms - sub-dividing waveforms associated with voice sounds into intervals corresp. to responses of vocal duct to series of excitation impulses of cords and synchronous to fundamental frequency of each signal

Patent Assignee: CSELT CENT STUDI LAB TELECOM SPA (CSEL)

Inventor: FOTI E; NEBBIA L; SANDRI S
 Number of Countries: 012 Number of Patents: 009
 Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 706170	A2	19960410	EP 95107944	A	19950524	199619	B
JP 8110789	A	19960430	JP 95175553	A	19950620	199627	
CA 2150614	A	19960330	CA 2150614	A	19950531	199628	
IT 1266943	B	19970121	IT 94TO756	A	19940929	199727	
EP 706170	A3	19971126	EP 95107944	A	19950524	199816	
ES 2113329	T1	19980501	EP 95107944	A	19950524	199824	
US 5774855	A	19980630	US 95528713	A	19950915	199833	
CA 2150614	C	20000411	CA 2150614	A	19950531	200035	
JP 3078205	B2	20000821	JP 95175553	A	19950620	200043	

Priority Applications (No Type Date): IT 94TO756 A 19940929

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 706170	A2	E	25	G10L-005/04	
Designated States (Regional): BE DE DK ES FR GB IT NL SE					
JP 8110789	A		16	G10L-003/00	
CA 2150614	A			G10L-009/00	
IT 1266943	B			G10L-000/00	
EP 706170	A3			G10L-005/04	
ES 2113329	T1			G10L-005/04	Based on patent EP 706170
US 5774855	A			G10L-009/12	
CA 2150614	C	E		G10L-009/00	
JP 3078205	B2		15	G10L-013/08	Previous Publ. patent JP 8110789
International Patent Class (Main): G10L-000/00; G10L-003/00; G10L-005/04; G10L-009/00; G10L-009/12; G10L-013/08					
International Patent Class (Additional): G10L-013/06					

Speech synthesis method using concatenation and partial overlapping of waveforms - ...

...sub-dividing waveforms associated with voice sounds into intervals corresp. to responses of vocal duct to series of excitation impulses of cords and synchronous to fundamental frequency of

...Abstract (Basic): The speech signal synthesis method involves using time-**concatenation** of **waveforms** representing elementary speech. The **waveforms** associated with voice sounds are sub-divided into intervals corresp. to responses the vocal duct to a series of impulses of vocal chord excitation and synchronous with the fundamental **waveform** frequency. The **waveform** in each interval is weighted, and the resulting signals are replaced with a replica shifted in time by an amount depending on **prosodic** information. The synthesis is performed by overlapping and adding the shifted signals...

...left and right analysis edges. Two connecting functions are applied in turn, and each interval of the synthesised signal is built by reproducing unchanged the **waveform** in the unchanging part of the original interval, and by aligning in time and adding the **waveforms generated** by the connecting functions...

...USE/ADVANTAGE - Pref. for text -to-speech synthesis. Synthesis signal has more **natural sound**.

...Title Terms: CONCATENATED ;

36/3,IC,K/10 (Item 10 from file: 350)
 DIALOG(R)File 350:Derwent WPIX

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010563524

WPI Acc No: 1996-060477/199607

XRPX Acc No: N96-050445

Text to speech system e.g. for workstation interaction, disabled person aid - controls operation of linguistic processor according to request signal from acoustic processor to process dispatcher indicating it is ready to process more speech segment from linguistic

Patent Assignee: INT BUSINESS MACHINES CORP (IBMC); IBM CORP (IBMC)

Inventor: SHARMAN R A

Number of Countries: 005 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
GB 2291571	A	19960124	GB 9414539	A	19940719	199607 B
EP 694904	A2	19960131	EP 95301164	A	19950222	199609
JP 8030287	A	19960202	JP 95122096	A	19950522	199615
EP 694904	A3	19971022	EP 95301164	A	19950222	199814
US 5774854	A	19980630	US 94343304	A	19941122	199833

Priority Applications (No Type Date): GB 9414539 A 19940719

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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GB 2291571	A		21	G10L-005/04	
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EP 694904	A2 E	12		G10L-005/04	
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Designated States (Regional): DE FR GB

JP 8030287	A	11		G10L-003/00	
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EP 694904	A3			G10L-005/04	
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US 5774854	A			G10L-005/02	
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International Patent Class (Main): G10L-003/00; G10L-005/02; G10L-005/04

International Patent Class (Additional): G06F-003/16; G10L-009/00

Text to speech system e.g. for workstation interaction, disabled person aid...

...Abstract (Basic): The TTS (text to speech) system converts input text into an output acoustic signal simulating natural speech. The system has a linguistic processor (210) for generating a listing of speech segments and associated parameters from the input text. An acoustic processor (220) generates the output acoustic waveform from this listing...

36/3,IC,K/11 (Item 11 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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008388294

WPI Acc No: 1990-275295/199036

XRPX Acc No: N90-212896

Text to speech synthesis system - has parameter generator that converts formant allophone data derived from code book tables

Patent Assignee: CENTIGRAM COMMUNICATIONS CORP (CENT-N); MALSHEEN B J (MALS-I); SPEECH PLUS INC (SPEE-N)

Inventor: GRONER G F; MALSHEEN B J; WILLIAMS L D; GRONER G; WILLIAMS L

Number of Countries: 015 Number of Patents: 006

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 9009657	A	19900823				199036 B
US 4979216	A	19901218	US 89312692	A	19890217	199102
EP 458859	A	19911204	EP 90903452	A	19900202	199149

EP 458859	A4	19920520	EP 90903452	A	19900000	199522
EP 458859	B1	19970730	EP 90903452	A	19900202	199735
			WO 90US528	A	19900202	
DE 69031165	E	19970904	DE 631165	A	19900202	199741
			EP 90903452	A	19900202	
			WO 90US528	A	19900202	

Priority Applications (No Type Date): US 89312692 A 19890217

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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WO 9009657	A				
					Designated States (National): CA JP
					Designated States (Regional): AT BE CH DE DK ES FR GB IT LU NL SE
EP 458859	A				
					Designated States (Regional): DE GB
EP 458859	B1	E	30	G10L-005/04	Based on patent WO 9009657
					Designated States (Regional): DE GB
DE 69031165	E			G10L-005/04	Based on patent EP 458859
					Based on patent WO 9009657

International Patent Class (Main): G10L-005/04

International Patent Class (Additional): G06F-015/34; G10L-005/00

Text to speech **synthesis system**...

...has parameter generator that converts formant allophone data derived from code book tables

...Abstract (Basic): The **text -to-speech** synthesiser reads the text and uses the spelling to **generate** phonemes where appropriate, but uses a dictionary look-up where the spelling is misleading. The consonant allophones are **generated** in the usual way but the vowels also have their allophones chosen by their context. All known allophones for a given language are stored in...

...ADVANTAGE - By choosing vowel as well as formant allophones the synthetic speech is made to **sound** more **natural** . (50pp Dwg.No.7/11)

...Abstract (Equivalent): A **text -to-speech** synthesis system, comprising: text conversion means (20, 22, 24) for converting a specified text string into a corresponding string of consonant and vowel phonemes (25), each the phoneme being selected from a predefined set of phonemes including a multiplicity of consonant phonemes and a multiplicity of vowel phonemes; parameter **generating** means (40) for **generating** speech parameters corresponding to the string of phonemes (25); and speech synthesising means (42) for **generating** a speech **waveform** corresponding to the speech parameters **generated** by the parameter **generating** means; characterised by: vowel allophone storage means (90, 130) storing a multiplicity of predefined vowel allophones, each vowel allophone being represented by a set of...

...and for then assigning to the vowel phoneme a selected one of the predefined vowel allophones corresponding to the computed phoneme context value; the parameter **generating** means (40) including means for **generating** speech parameters for the assigned vowel allophones...

...Abstract (Equivalent): The **text -to-speech** conversion system has a parameter **generator** which converts the phonemes into formant parameters, and a formant synthesiser which uses the formant parameters to **generate** a synthetic speech **waveform** . A library of vowel allophones are stored each stroed vowel allophone being represented by formant parameters for four formants. The vowel allophone library includes a...

...allophone with one or more pairs of phonemes preceding and following the

corresponding vowel phoneme in a phoneme string. When synthesising speech, a vowel allophone **generator** uses the vowel allophone library to provide formant parameters representative of a specified vowel phoneme. The vowel allophone **generator** coacts with the context index to select the proper vowel allophone, as determined by the phonemes preceding and following the specified vowel phoneme. ADVANTAGE - Synthesised...

...Title Terms: **GENERATOR** ;

36/3,IC,K/12 (Item 12 from file: 350)
 DIALOG(R)File 350:Derwent WPIX
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008229149

WPI Acc No: 1990-116150/199015

XRPX Acc No: N90-089960

Waveform **addition-overlapping speech synthesis - using dictionary of diphone sound element derived by window analysis of speech signal**

Patent Assignee: FRANCE TELECOM (ETFR); ETAT FR MIN PTT (ETFR); MIN POSTS TELECOM & SPACE CENT NAT ETUD (ETFR); HAMON C (HAMO-I)

Inventor: HAMON C

Number of Countries: 007 Number of Patents: 011

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 9003027	A	19900322				199015	B
EP 363233	A	19900411	EP 89402394	A	19890901	199015	
FR 2636163	A	19900309				199017	
DK 9001073	A	19900530				199040	
JP 3501896	W	19910425	JP 89509621	A	19890901	199123	
CA 1324670	C	19931123	CA 610127	A	19890901	199402	
US 5327498	A	19940705	WO 89FR438	A	19890901	199426	
			US 90487942	A	19901115		
EP 363233	B1	19941130	EP 89402394	A	19890901	199501	
DE 68919637	E	19950112	DE 619637	A	19890901	199507	
			EP 89402394	A	19890901		
ES 2065406	T3	19950216	EP 89402394	A	19890901	199513	
US 5524172	A	19960604	WO 89FR438	A	19890901	199628	
			US 90487942	A	19901115		
			US 94224652	A	19940404		

Priority Applications (No Type Date): FR 8811517 A 19880902

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
CA 1324670	C	F		G10L-005/04	
US 5327498	A		10	G10L-005/00	Based on patent WO 9003027
EP 363233	B1	F	12	G10L-005/04	
DE 68919637	E			G10L-005/04	Based on patent EP 363233
ES 2065406	T3			G10L-005/04	Based on patent EP 363233
US 5524172	A		9	G10L-005/04	Cont of application WO 89FR438
					Cont of application US 90487942
					Cont of patent US 5327498

EP 363233 A G10L-005/04

International Patent Class (Main): G10L-005/00; G10L-005/04

International Patent Class (Additional): G10L-003/02; G10L-009/00

Waveform **addition-overlapping speech synthesis...**

...Abstract (Equivalent): replaced with a time shift thereof equal to a fundamental synthesis period, which is lesser than or greater than the original fundamental period, responsive to **prosodic** information

relating to the fundamental synthesis frequency, (c) synthesis is carried out by summing the thus shifted signals, characterised in that the method does not...

...Abstract (Equivalent): element with a time shift thereof equal to the fundamental synthesis period, which is lesser than or greater than the original fundamental period responsive to **prosodic** information relative to the fundamental synthesis period, and...

...c) summing the thus shifted signal to **synthesize** speech, said method being devoid of a modification of a **pitch** period of the speech sounds elements by spectral transformation between steps (a) and (b)...

...The process comprises supplying a sequence of phoneme codes and respective **prosodic** information, and, for each phoneme, analysing and synthesising each phoneme, and then **concatenating** the **synthesized** phonemes. For each phoneme, two diphones are selected among the stored diphones and the presence of voicing is determined...

...For voiced phonemes, the respective **waveforms** of the two diphones constituting the phoneme are filtered by a window which is centered on a point of the selected **waveform** representative of the beginning of a pulse response of vocal cords to excitation. The window has a width substantially equal to twice the greater of...

...USE - Speech synthesis process using diphones stored in a dictionary as **waveforms**, for **text -to-speech** conversion...

Title Terms: **WAVEFORM** ;

36/3,IC,K/13 (Item 13 from file: 350)
DIALOG(R)File 350:Derwent WPIX
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007865050

WPI Acc No: 1989-130162/198917

XRPX Acc No: N89-099196

Generating **speech from digitally stored co-articulated speech segments**
- **recovering stored segments and concatenating in real time then**
applying data to sound generator

Patent Assignee: KANDEFER E M (KAND-I); SOUND ENTERTAINMENT INC (SOUN-N);
SOUND ENTERTAINMENT (SOUN-N)

Inventor: KANDEFER E M; MOSENFELDER J R; MOSENFELDE J R

Number of Countries: 018 Number of Patents: 012

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 8903573	A	19890420	WO 88US3479	A	19881007	198917	B
AU 8825481	A	19890502				198932	
NO 9001473	A	19900528				199027	
EP 380572	A	19900808	EP 88909070	A	19881007	199032	
JP 3504897	W	19911024	JP 88508356	A	19881007	199149	
US 5153913	A	19921006	US 89382675	A	19890619	199243	
AU 9221056	A	19921112	AU 9221056	A	19920814	199301	
			AU 8825481	A	19880000		
EP 380572	B1	19940727	EP 88909070	A	19881007	199429	
			WO 88US3479	A	19881007		
DE 3850885	G	19940901	DE 3850885	A	19881007	199434	
			EP 88909070	A	19881007		
			WO 88US3479	A	19881007		
AU 652466	B	19940825	AU 9221056	A	19920814	199436	
			AU 8825481	A	19880000		
EP 380572	A4	19910417	EP 88909070	A	19880000	199516	

CA 1336210 C 19950704 CA 579709 A 19881011 199534

Priority Applications (No Type Date): US 87107678 A 19871009

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 8903573 A E 47

Designated States (National): AU DK JP KR NO US

Designated States (Regional): AT BE CH DE FR GB IT LI LU NL SE

EP 380572 A

Designated States (Regional): CH DE FR GB IT LI NL SE

US 5153913 A 21 G10L-005/01

AU 9221056 A G01L-005/04 Div ex application AU 8825481

EP 380572 B1 E 26 G10L-005/04 Based on patent WO 8903573

Designated States (Regional): CH DE FR GB IT LI NL SE

DE 3850885 G G10L-005/04 Based on patent EP 380572

Based on patent WO 8903573

AU 652466 B G01L-005/04 Div ex application AU 8825481

Previous Publ. patent AU 9221056

CA 1336210 C G10L-005/04

International Patent Class (Main): G01L-005/04; G10L-005/01; G10L-005/04

International Patent Class (Additional): G01L-009/18

Generating **speech from digitally stored co-articulated speech segments**

...

...recovering stored segments and concatenating in real time then
applying data to sound generator

...Abstract (Basic): beginning, ending, and intermediate diphone sounds
from the recorded syllables. Data samples are stored representing the
extracted sounds in a digital memory device. A selected **text** to
speech sequence of diphones required to **generate** a desired message
is **generated** .

...

...Stored data is recovered from the digital memory for each diphone in the
selected sequence. The selected sequence of diphones is **concatenated**
directly without any interpolation signals, in real time, using the
recovered data. The **concatenated** diphone data is applied to a sound
generating circuit to **generate** a desired message with a 3 KHz
bandwidth...

...ADVANTAGE - Quality speech is **generated** using a reduced amount of
storage space and speech segments are joined in real time with smooth
transitions required for quality speech.

...Abstract (Equivalent): A method of **generating** speech using prerecorded
real speech diphones, said method comprising the steps of: digitally
recording as PCM data samples spoken carrier syllables in which desired
diphones...

...the PCM data samples representing desired beginning, ending and
intermediate diphones from the digitally recorded carrier syllables at
a substantially common preselected location in the **waveform** of each
diphone; digitally compressing (27-85) the PCM samples of said
diphones using adaptive differential pulse code modulation to
generate ADPCM encoded data; storing (77) the ADPCM encoded data
representing said extracted digital diphones in a digital memory
devices (91); **generating** (95) a selected **text** to **speech** sequence
of diphones required to **generate** a desired message; recovering (115)
stored ADPCM encoded data from said digital memory device (91) for each
diphone in said selected sequence of diphones; reconstructing (123)

the PCM diaphone data samples from said recovered ADPCM encoded data; **concatenating** said reconstructed PCM diaphone data samples in said selected **text** to **speech** sequence of diaphones coarticulated speech segments directly, in real time; and applying (125) the **concatenated** reconstructed diaphone data **samples** to **sound generating** means (97-101) to **generate** said desired message; said method characterised by compressing the PCM data samples by **generating** (27, 31) a seed quantiser for the first data sample in each diaphone, by storing (29, 33) the seed quantiser for the first data sample...

...Abstract (Equivalent): are extracted from spoken carrier syllables and digitally compressed for storage using adaptive differential pulse code modulation (ADPCM). Beginning seed quantization and PCM values are **generated** for each coarticulated speech segment and stored together with the ADPCM encoded data in a coarticulated speech segment library ...

...ADPCM encoded data are recovered from the coarticulated speech segment library and blown back using the initial quantization and PCM seed values. This reconstructs and **concatenates** in real time the sequence of coarticulated speech segments required by a **text** to **speech** program to **generate** a desired high quality spoken message. Pref. the coarticulated speech segments are diphones...

...USE - **Generating** quality speech from prerecorded digitally stored spoken speech segments in library in real time. Reduced memory requirements. (Dwg.10/10

Title Terms: **GENERATE** ;

36/3,IC,K/14 (Item 1 from file: 347)

DIALOG(R)File 347:JAPIO

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05866088

TEXT VOICE SYNTHESIZER

PUB. NO.: 10-149188 [JP 10149188 A]

PUBLISHED: June 02, 1998 (19980602)

INVENTOR(s): FUJIMOTO HIROYUKI

YAMATO TOSHITAKA

ISHIKAWA OSAMU

APPLICANT(s): FUJITSU TEN LTD [421134] (A Japanese Company or Corporation),
JP (Japan)

APPL. NO.: 08-310890 [JP 96310890]

FILED: November 21, 1996 (19961121)

INTL CLASS: [6] G10L-003/00; G10L-005/02

TEXT VOICE SYNTHESIZER

ABSTRACT

PROBLEM TO BE SOLVED: To form almost natural voice **synthesization** concerning limited sentence examples...

...SOLUTION: Concerning a **text voice synthesizer** for regularly **synthesizing** arbitrary sentences in voice, this device is provided with a word dictionary part 62 storing a lot of words and having identification attributes in partial...

... control part 65 for collating a word string provided from a language processing analytic part 63 with the sentence example pattern and controlling inserted voice **synthesization** or regular **synthesization** . A

device for performing the inserted voice **synthesization** is provided with an insert table 73 with **intonation** composed of conjugation for conjugating plural words and the **intonations** of inserted word strings and an inserted rhythm **generating** part 74 for **generating** a **pitch** pattern while using the **intonation** of inserted word string and connecting a **waveform** for the unit of a voice according to this **pitch** pattern.

36/3,IC,K/15 (Item 2 from file: 347)
DIALOG(R)File 347:JAPIO
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05814190

SPEECH SYNTHESIZER

PUB. NO.: 10-097290 [JP 10097290 A]
PUBLISHED: April 14, 1998 (19980414)
INVENTOR(s): NISHIDA HIDEJI
HIRAI HIROYUKI
MIYATAKE MASANORI
ONISHI HIROKI
APPLICANT(s): SANYO ELECTRIC CO LTD [000188] (A Japanese Company or Corporation), JP (Japan)
APPL. NO.: 08-251646 [JP 96251646]
FILED: September 24, 1996 (19960924)
INTL CLASS: [6] G10L-005/04; G10L-003/00

SPEECH SYNTHESIZER

ABSTRACT

PROBLEM TO BE SOLVED: To output a **synthesized** speech **waveform** of superior speech quality by reading an optimum unit speech **waveform** corresponding to a 1st vocal sound symbol part string divided in specific preferential order out of a **waveform** memory and connecting it...

...SOLUTION: A **text speech synthesizer** 10 includes a microcomputer 12. The microcomputer 12 receives an input character string consisting of a 1st vocal sound symbol string consisting of text document...

... dictionary 14 for text analysis to convert it into a vocal sound symbol string consisting of the 1st vocal sound symbol part string and also **generate** the **pitch** pattern and power pattern of this input character string. Then the microcomputer 12 shapes, connects, and edits unit speech **waveforms** registered in a speech **waveform** data base 16 according to the **pitch** pattern and power pattern, and outputs the resulting **synthesized** speech. Language information corresponding to vocal sound symbols of a 2nd vocal sound symbol string which is divided in specific preferential order is added to...

36/3,IC,K/16 (Item 3 from file: 347)
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05279284

HARMONY GENERATING DEVICE

PUB. NO.: 08-234784 [JP 8234784 A]
PUBLISHED: September 13, 1996 (19960913)
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APPLICANT(s): YAMAHA CORP [000407] (A Japanese Company or Corporation), JP
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APPL. NO.: 07-041767 [JP 9541767]
FILED: March 01, 1995 (19950301)
INTL CLASS: [6] G10K-015/04; G10H-001/38

HARMONY **GENERATING** DEVICE

ABSTRACT

PURPOSE: To provide a KARAOKE device which **generates** a harmony voice signal even unless the **pitch** of a **text voice** signal is detected...

... sound volume detection part 43, and a multiplier 45. The singing voice signal is multiplied by a window function through the multiplier 45 and cut **waveform** element data of one cycle are stored in a memory 46. A readout control part 48 for harmony data accesses the memory 46 and the signal obtained by repeatedly reading **waveform** element data out at intervals corresponding to a harmony frequency is the harmony voice signal. The window function is one cycle long in terms of...

... window function so controlled that the peak detected by the peak detection part 41 is at the center of the window function. A window function **generation** part 44 cuts the **waveform** element data at intervals of tens of ms and **waveform** element data corresponding to a timbre are written in the memory 46: when phonemes change, a phoneme detection part 42 transmits that to the window function **generation** part 44 to **generate** the window function.

36/3,IC,K/17 (Item 4 from file: 347)
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03708597

DEVICE AND METHOD FOR **SYNTHESIZING** SOUND RULE

PUB. NO.: 04-073697 [JP 4073697 A]
PUBLISHED: March 09, 1992 (19920309)
INVENTOR(s): TAKEDA SHOICHI
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APPLICANT(s): HITACHI LTD [000510] (A Japanese Company or Corporation), JP
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APPL. NO.: 02-183947 [JP 90183947]
FILED: July 13, 1990 (19900713)
INTL CLASS: [5] G10L-005/00; G10L-003/00
JOURNAL: Section: P, Section No. 1374, Vol. 16, No. 276, Pg. 165, June 19, 1992 (19920619)

DEVICE AND METHOD FOR **SYNTHESIZING** SOUND RULE

ABSTRACT

PURPOSE: To realize increased or decreased intensity included in natural **text voice** vocalized by a person in rule synthesis by **synthesizing** voice sequentially by a phoneme parameter string in accordance with an input text and the time- changed pattern(**pitch** pattern) of a fundamental frequency...

...CONSTITUTION: A control parameter **generating** part 3 decides accent, **intonation**, phoneme **duration**, and a sound source power(amplitude) correction value by a rule, and **generates** the **pitch** pattern and a

phoneme parameter time series according to them. **Generated** fundamental frequency and phoneme parameter are sent to a voice synthesis part 4 sequentially, and a voice **waveform** is outputted. Thereby, since rhythm control by a prominence **generation** rule is found based on the quantitative analysis of natural voice, natural increased or decreased intensity looking like a human being can be supplied to the voice **synthesized** from an input document(text).

?